Avaya Solution Interoperability Lab

Configuring 9600-Series SIP Phones with Avaya Aura™ Session Manager Release 5.2 – Issue 1.0

Abstract

These Application Notes describe the configuration of 9600-Series SIP Phones with Avaya Aura™ Session Manager and Avaya Aura™ Communication Manager as a Feature Server.

- Avaya Aura™ Session Manager provides SIP proxy/routing functionality, routing SIP sessions across a TCP/IP network with centralized routing policies and registrations for SIP endpoints.
- Avaya Aura™ Communication Manager operates as a Feature Server for the SIP endpoints which communicate with Avaya Aura™ Session Manager over SIP trunks.

These Application Notes provide information for the setup, configuration, and verification of the call flows tested on this solution.

Table of Contents:

1. Introduction

These Application Notes present a sample configuration for a network that uses Avaya Aura™ Session Manager to support registration of 9600-Series SIP phones and enables connectivity to an Avaya Aura™ Communication Manager Feature Server 5.2.1 using SIP trunks.

As shown in **Figure 1,** Avaya Aura™ Session Manager is managed by Avaya Aura™ System Manager. Avaya 9620 IP Telephones configured as SIP endpoints utilize the Avaya Aura™ Session Manager User Registration feature and require an Avaya Aura™ Communication Manager operating as a Feature Server. Communication Manager Feature Server only supports IP Multimedia Subsystem (IMS)-SIP users that are registered to Avaya Aura™ Session Manager. The Communication Manager Feature Server is connected to Session Manager via an IMS-enabled SIP signaling group and associated SIP trunk group.

The Avaya 9600-Series IP Telephone (H.323) and 2420 Digital Telephone are supported by Avaya Aura™ Communication Manager Access Element. The Communication Manager Access Element is connected over a SIP trunk to the Avaya Aura™ Session Manager, using its SM-100 (Security Module) network interface. All inter-system calls are carried over these SIP trunks.

For the sample configuration, Avaya Aura™ Session Manager runs on an Avaya S8510 Server, Avaya Aura™ Communication Manager 5.2.1 Feature Server runs on a S8300 Server with Avaya G450 Media Gateway, and Avaya Aura™ Communication Manager 5.2.1 Access Element runs on an Avaya S8730 Server with Avaya G650 Media Gateway. The results in these Application Notes should be applicable to other Avaya servers and media gateways that support Avaya Aura™ Communication Manager 5.2.1.

These Application Notes will focus on the configuration of the Communication Manager Feature Server and Session Manager. Detailed administration of Communication Manager Access Element will not be described (see the appropriate documentation listed in **Section 8**)**.**

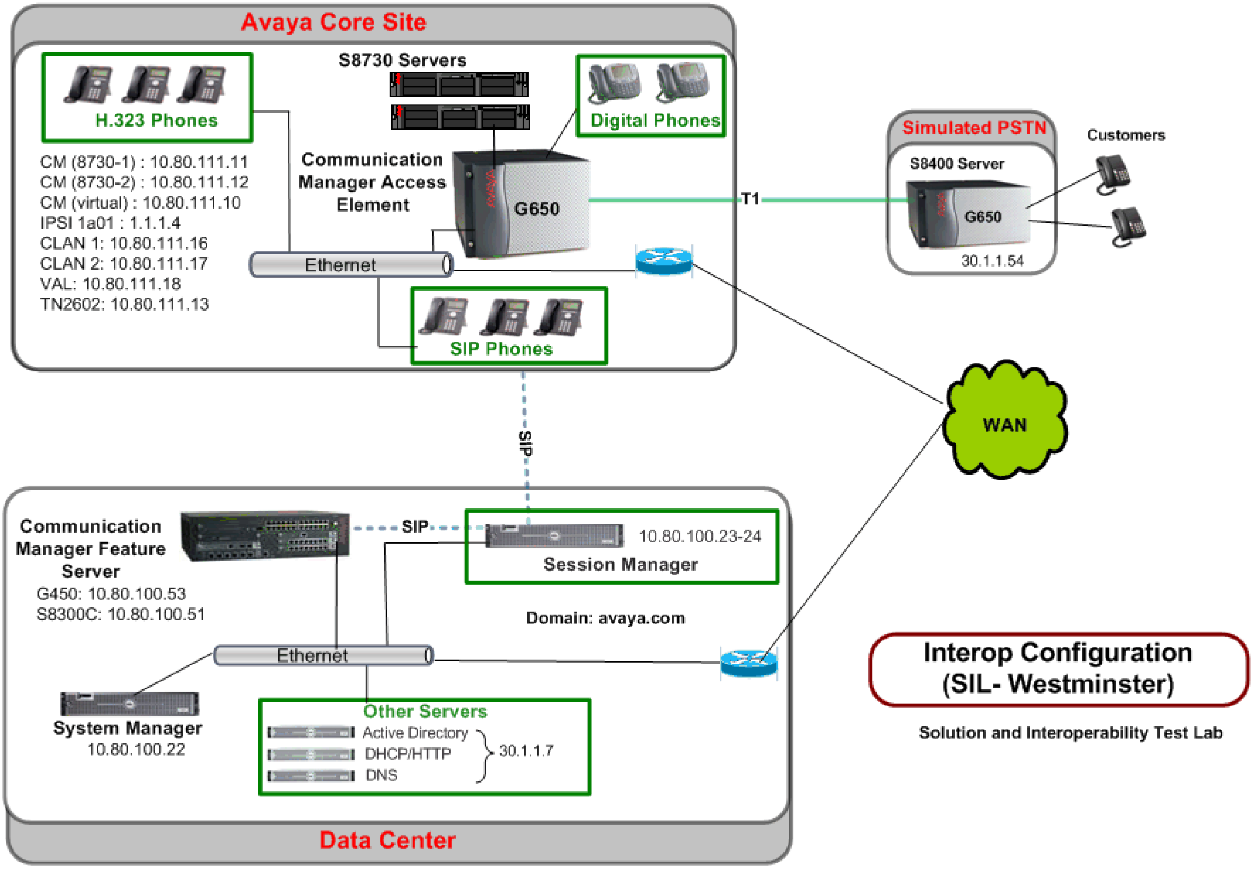


Figure 1 – Sample Configuration

1.1. Equipment and Software Validated

The following equipment and software were used for the sample configuration.

Equipment	Software
Avaya Aura^{TM} Session Manager	Release 5.2.0.1.520017-11-18-2009
Avaya Aura^{TM} System Manager	Release 5.2, Load: 5.2.0.8.27
Avaya Aura^{TM} Communication Manager • Avaya S8730 Server Access Element	5.2.1 R015x.02.1.016.4
Avaya Aura^{TM} Communication Manager • Avaya S8300 Feature Server	5.2.1 R015x.02.1.016.4
Avaya IP Telephones (H.323): • 9650 • 9630 • 9620	 FW: 2.0 FW: 3.0 FW:1.5
Avaya SIP Phones • 9630	FW: 2.5.5.16
Avaya Digital Telephones (8410D)	N/A

2. Configuring Avaya Aura™ Communication Manager Feature Server

This section describes the administration of Communication Manager Feature Server using a System Access Terminal (SAT). Alternatively, some of the station administration could be performed using the Communication System Management application on System Manager. These instructions assume the G450 Media Gateway is already configured on the Communication Manager Feature Server. Some administration screens have been abbreviated for clarity.

- Verify System Capabilities and Communication Manager Licensing
- Administer network region
- Administer IP node names
- Administer IP interface
- Administer SIP trunk group and signaling group
- Administer route patterns
- Administer numbering plan

After completing these steps, the "**save translations**" command should be performed.

2.1. Verify System Capabilities and Licensing

This section describes the procedures to verify the correct system capabilities and licensing have been configured. If there is insufficient capacity or a required feature is not available, contact an authorized Avaya sales representative to make the appropriate changes.

2.1.1. SIP Trunk Capacity Check

Issue the **display system-parameters customer-options** command to verify that an adequate number of SIP trunk members are licensed for the system as shown below:

```
display system-parameters customer-options                Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 500   0
           Maximum Concurrently Registered IP Stations: 18000 4
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 50    20
```

2.1.2. AAR/ARS Routing Check

Verify that **ARS** and **ARS/AAR Dialing without FAC** are enabled (on page 3 of system-parameters customer options).

```
display system-parameters customer-options                    Page   3 of  11
                                OPTIONAL FEATURES

 A/D Grp/Sys List Dialing Start at 01? n                             CAS Main? n
Answer Supervision by Call Classifier? n                    Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                        DCS (Basic)? y
          ASAI Link Core Capabilities? y                    DCS Call Coverage?
```

2.1.3. Configure Trunk-to-Trunk Transfers

Use the "**change system-parameters features**" command to enable trunk-to-trunk transfers. This feature is needed to be able to transfer an incoming/outgoing call from/to the remote switch back out to the same or another switch For simplicity, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution.

```
change system-parameters features                               Page   1 of  18
                                FEATURE-RELATED SYSTEM PARAMETERS
                                   Self Station Display Enabled? n
                                        Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
```

2.1.4. Enable Private Numbering

Use the "**change system-parameters customer-options**" command to verify that Private Networking is enabled as shown below:

```
display system-parameters customer-options                 Page   5 of  11
                                OPTIONAL FEATURES

               Multinational Locations? y            Station and Trunk MSP? y
Multiple Level Precedence & Preemption? n     Station as Virtual Extension? y
                    Multiple Locations? y
                                           System Management Data Transfer? n
         Personal Station Access (PSA)? y              Tenant Partitioning? n
                       PNC Duplication? n      Terminal Trans. Init. (TTI)? y
                  Port Network Support? n              Time of Day Routing? n
                       Posted Messages? n      TN2501 VAL Maximum Capacity? y
                                                      Uniform Dialing Plan? y
                    Private Networking? y    Usage Allocation Enhancements? y
              Processor and System MSP? y
                    Processor Ethernet? y              Wideband Switching? n
                                                                 Wireless? y
```

2.2. Add Node Name of Avaya Aura™ Session Manager

Using the **change node-names ip** command, add the node-name and IP for the Session Manager, if not previously added.

```
change node-names ip                                       Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
ASM1                10.80.100.24
Nortel-CS1000e      10.80.50.50
default             0.0.0.0
procr               10.80.100.51
```

2.3. Configure IP Network Region

Using the **change ip-network-region 1** command, set the **Authoritative Domain** to the correct SIP domain for the configuration. Verify the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio** fields are set to "**yes**".

```
change ip-network-region 1                                 Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1        Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 16585
```

2.4. Add SIP Signaling Group

Issue the **add signaling-group n** command, where "n" is an available signaling group number, for one of the SIP trunks to the Session Manager, and fill in the indicated fields.

In the sample configuration, trunk group “10” and signaling group “10” were used to connect to Avaya Aura™ Session Manager. Default values can be used for the remaining fields.

- **Group Type:** “sip”
- **Transport Method:** “tcp[1]”
- **IMS Enabled?:** “y”
- **Near-end Node Name:** procr from **Section 2.2**
- **Far-end Node Name:** Session Manager node name from **Section 2.2**
- **Near-end Listen Port:** “5060”
- **Far-end Listen Port:** “5060”
- **Far-end Domain:** Authoritative Domain from **Section 2.3**
- **Enable Layer 3 Test:** “y”
- **Session Establishment Timer:** “3”[2]

```
display signaling-group 10                                    Page   1 of   1
                                SIGNALING GROUP

Group Number: 10             Group Type: sip
                        Transport Method: tcp
  IMS Enabled? y
     IP Video? n

    Near-end Node Name: procr                     Far-end Node Name: ASM1
 Near-end Listen Port: 5060                     Far-end Listen Port: 5060
                                            Far-end Network Region:  1
Far-end Domain: avaya.com
                                            Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                 RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                      IP Audio Hairpinning? n
         Enable Layer 3 Test? y                     Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n         Alternate Route Timer(sec):  6
```

2.5. Add SIP Trunk Group

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where “n” is an available trunk group number and fill in the indicated fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”

[1] TCP was used for the sample configuration. However, TLS would typically be used in production environments.
[2] If any call originating from the SIP phone is not expected to be answered within 3 minutes such would happen if the call is made to a VDN and agents are not available within 3 minutes, this value may need to be increased.

- **Signaling Group:** The number of the signaling group added in **Section 2.4**
- **Number of Members:** The number of SIP trunks to be allocated to calls routed to Session Manager (must be within the limits of the total number of trunks configured in **Section 2.1.1**).

```
add trunk-group 10                                                  Page   1 of  21
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: ASM1                        COR: 1        TN: 1        TAC: #10
Direction: two-way        Outgoing Display? n
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                     Signaling Group: 10
                                                   Number of Members: 10
```

Once the add command is completed, trunk members will be automatically generated based on the value in the **Number of Members** field.

On page 2, set the **Preferred Minimum Session Refresh Interval** to 1200. Note: to avoid extra SIP messages, all SIP trunks connected to Session Manager should be configured with a minimum value of 1200.

```
add trunk-group 10                                                  Page   2 of  21
                                     Group Type: sip

TRUNK PARAMETERS

     Unicode Name: auto

                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                               Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1200
```

On page 3, set **Numbering Format** to be *private*. Use default values for all other fields.

```
add trunk-group 10                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                      Maintenance Tests? y




                     Numbering Format: private
                                                UUI Treatment: service-provider

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n
```

2.6. Administering Numbering Plan

SIP Users registered to Session Manager need to be added to either the private or public numbering table on the Communication Manager Feature Server. For the sample configuration, private numbering was used and all extension numbers were unique within the private network. However, in many customer networks, it may not be possible to define unique extension numbers for all users within the private network. For these types of networks, additional administration may be required as described in References [3] and [8].

To enable SIP endpoints to dial extensions defined in the Communication Manager Access Element, use the "**change private-numbering x"** command, where x is the number used to identify the private number plan. For the sample configuration, extension numbers starting with 5XX-XXXX or 6XX-XXX are used on the Communication Manager Access Element.

- **Ext Len:** Enter the extension length allowed by the dial plan
- **Ext Code:** Enter leading digit (s) from extension number
- **Trunk Grp:** Enter the SIP Trunk Group number for the SIP trunk between the Feature Server and Session Manager
- **Private Prefix:** Leave blank unless an enterprise canonical numbering scheme is defined in Session Manager. If so, enter the appropriate prefix.

```
change private-numbering 1                                    Page   1 of  2
                            NUMBERING - PRIVATE FORMAT

Ext Ext               Trk         Private          Total
Len Code              Grp(s)      Prefix           Len
 7  5                 10                           7     Total Administered: 2
 7  6                 10                           7        Maximum Entries: 540
```

2.7. Configure Stations

For each SIP user to be defined in Session Manager, add a corresponding station on the Communication Manager Feature Server. Note: instead of manually defining each station using the Communication Manager SAT interface, an alternative option is to automatically generate the SIP station when adding a new SIP user. See Section 3.4.6 for more information on adding SIP users.

The phone number defined for the station will be the number the SIP user enters to register to Session Manager. Use the "**add station x**" command where x is a valid extension number defined in the system. On page 1 of the change station form:

- **Phone Type:** Set to 96xxSIP
- **Name:** Display name for user
- **Security Code:** number used when user logs into station. Note: this code should match the "**Shared Communication Profile Password**" field defined when adding this user in Session Manager. See **Section 3.4.5**.

```
add station 6663000                                 Page   1 of   6
                             STATION

Extension: 666-3000                      Lock Messages? n          BCC: 0
     Type: 9630SIP                       Security Code: 123456      TN: 1
     Port: S00006                      Coverage Path 1: 1          COR: 1
     Name: John Smith                  Coverage Path 2:            COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19
                                                 Message Lamp Ext: 666-3000
        Display Language: english                  Button Modules: 0
          Survivable COR: internal
   Survivable Trunk Dest? y                          IP SoftPhone? n

                                                         IP Video? n
```

On page 6, set:

- **SIP Trunk option:** Enter SIP Trunk Group defined in **Section 2.5**

```
change station 6663000                                 Page   6 of   6
                             STATION
SIP FEATURE OPTIONS
         Type of 3PCC Enabled: None
                    SIP Trunk: 10
```

Note: an alternative option for configuring stations is to use the option when adding a SIP user in Session Manager to automatically generate the station. **See Section 3.4.5** for more information on using Session Manager to add SIP users.

2.8. Configure Off-PBX-Telephone Station-Mapping

Use the "**change off-pbx-telephone station-mapping**" command for each extension associated with SIP users defined in Session Manager. On page 1, enter the SIP Trunk Group defined in **Section 2.5** and use default values for other fields.

```
change off-pbx-telephone station-mapping 6663000              Page   1 of   3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station        Application Dial   CC  Phone Number    Trunk      Config  Dual
 Extension                  Prefix                     Selection  Set     Mode
 666-3000          OPS           -     6663000          10         1
                                 -
                                 -
```

On page 2, enter the following values:

- **Mapping Mode**: "both"
- **Calls Allowed:** "all"

```
change off-pbx-telephone station-mapping 6663000              Page   2 of   3

                STATIONS WITH OFF-PBX TELEPHONE INTEGRATION


 Station        Appl    Call         Mapping    Calls      Bridged     Location
 Extension      Name    Limit        Mode       Allowed    Calls
 666-3000       OPS      3           both       all         none
                                 -
```

2.9. Save Translations

Configuration of Communication Manager Feature Server is complete. Use the **"save translations**" command to save these changes

Note: After a change on Communication Manager Feature Server which alters the dial plan, synchronization between Communication Manager Feature Server and Session Manager needs to be completed and SIP phones must be re-registered. To request an on demand synchronization, log into the System Manager console and use the **Synchronize CM Data** feature under the Communication System Management menu.

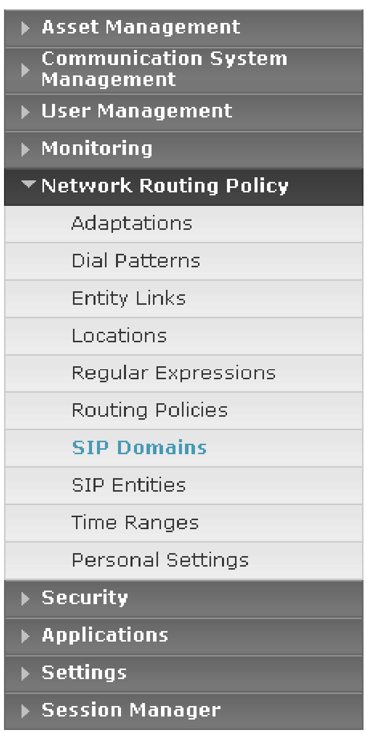
3. Configure Avaya Aura™ Session Manager

This section provides the procedures for configuring the Session Manager and includes the following items:

- Administer SIP domain
- Define Logical/physical Locations that can be occupied by SIP Entities
- For each SIP entity in the sample configuration:
    - o Define SIP Entity
    - o Define Entity Links, which define the SIP trunk parameters used by Avaya Aura ™ Session Manager when routing calls to/from SIP Entities
    - o Define Routing Policies, which control call routing between the SIP Entities
    - o Define Dial Patterns, which govern to which SIP Entity a call is routed
- Define the Communication Manager Feature Server as an administration entity
- Adding SIP Endpoints/SIP URE users

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura™ System Manager, using the URL “http://<ip-address>/SMGR”, where “<ip-address>” is the IP address of Avaya Aura™ System Manager.

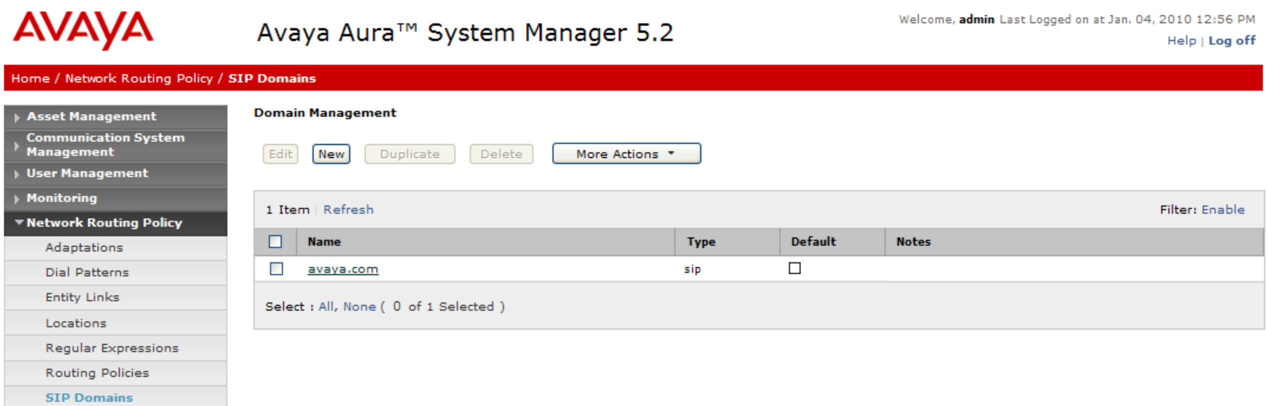
Log in with the appropriate credentials and accept the Copyright Notice.
Expand the **Network Routing Policy** Link on the left side of Navigation Menu. Select a specific item such as SIP Domains. When the specific item is selected, the color of the item will change to blue as shown below:



3.1. Administer SIP Domains

- Expand Network Routing Policy and select **SIP Domains**.
  - Click **New**
  - In the *General* Section, under *Name* add a descriptive name. Under *Notes* add a brief description.
  - Click **Commit** to save.

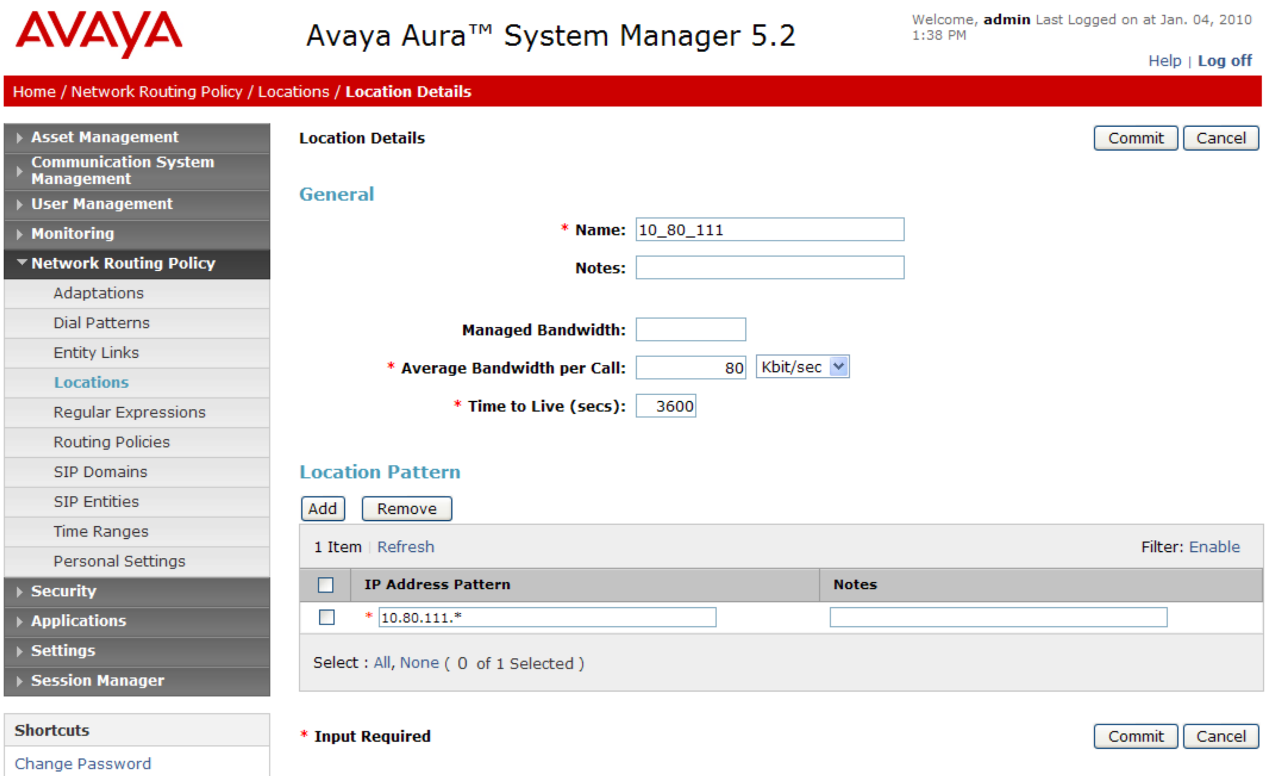
The screen below shows the information for the sample configuration.



3.2. Define Locations

- Expand Network Routing Policy and select **Locations.** Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing.
  - Click **New**
  - In the *General* Section, under *Name* add a descriptive name.
  - Under *Notes* add a brief description.
  - In the *Location Pattern* Section, under IP Address Pattern enter pattern used to logically identify the location Under *Notes* add a brief description.
  - Click **Commit** to save.

The screen below shows the information for Communication Manager Access Element in the sample configuration.

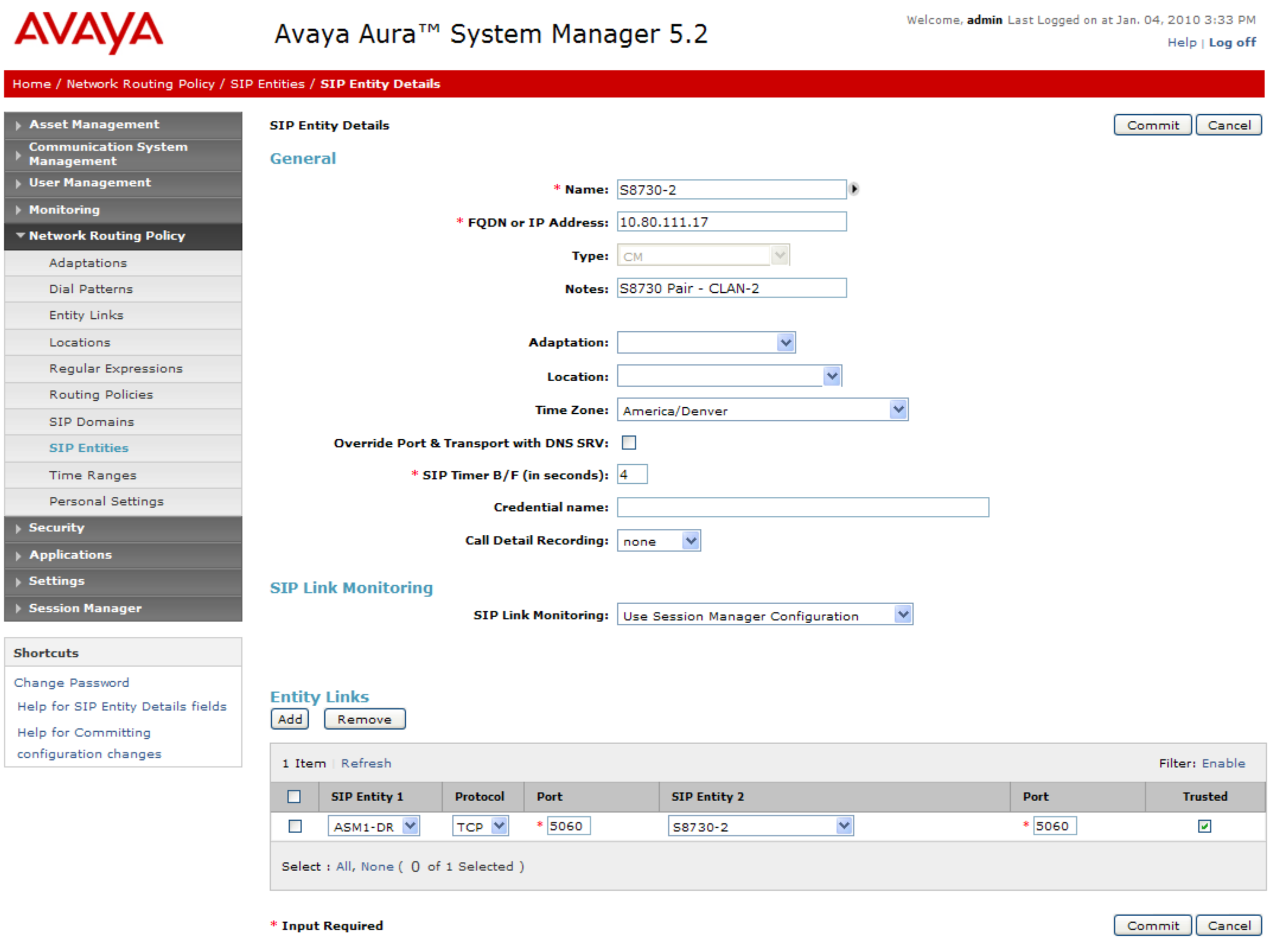


3.3. Add Avaya Aura™ Communication Manager Access Element

3.3.1. Define SIP Entity for the Avaya Aura™ Communication Manager Access Element

- Expand Network Routing Policy
  - o Select SIP Entities
    - Click **New**
    - In the *General* Section, under *Name* add an identifier for the Communication Manager. Under *FQDN or IP Address* enter the IP Address of the Communication Manager. Under *Type* select CM. Under *Notes* add a brief description.
    - *Location:* From the drop-down select the Location added in **Section 3.2**. Note: since location-based routing was not used in the sample configuration, selecting a value for location field is optional.
    - Click **Commit** to save.

The following screen shows addition of Communication Manager Access Element. The IP address used is that of the C-LAN board in the Avaya G650 Media Gateway.

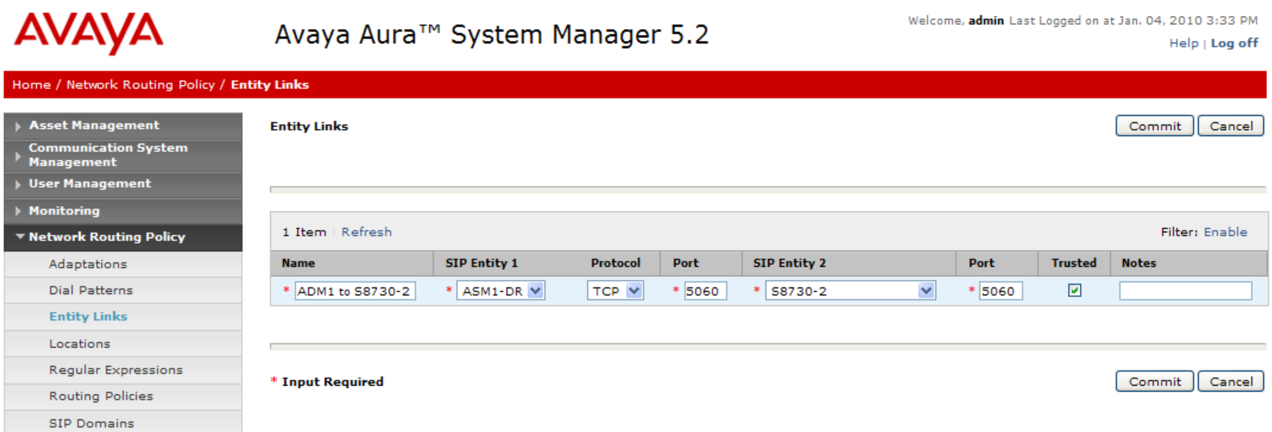


3.3.2. Define an Entity Link for Avaya Aura™ Communication Manager Access Element

- Expand Network Routing Policy
  - o Entity Links
    - Click **New**
    - Under *Name,* enter an identifier for the Communication Manager Access Element.
    - Under *SIP Entity 1* drop-down select the appropriate Session Manager. Under *Port* drop-down select the correct port for the Session Manager.
    - Under *SIP Entity 2* drop-down select the SIP Entity added in **Section 3.3.1** for the Communication Manager Access Element. Under *Port* drop-down select the correct port for the Communication Manager. Select it as a *Trusted* host. Under *Protocol* drop-down select the required protocol.
    - Under *Notes* add a brief description.

- Click **Commit** to save.

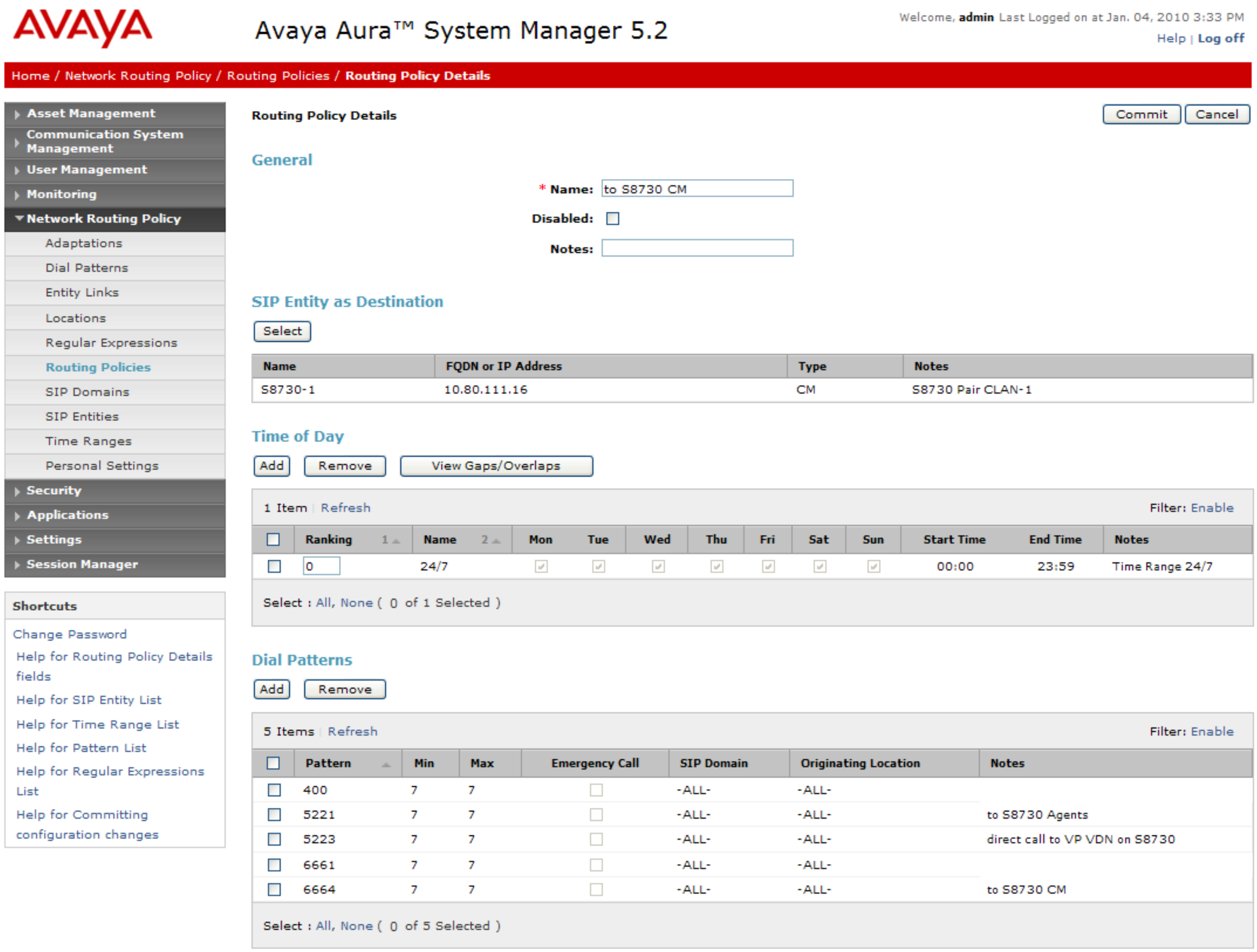
The following screen shows the entity link defined for the Communication Manager Access Element.



3.3.3. Define Routing Policy for Avaya Aura™ Communication Manager Access Element

- Expand Network Routing Policy
  - o Routing Policies
    - Click **New**
    - In the 'General' section, under Name add an identifier to define the routing policy for the Communication Manager. Under *Notes* add a brief description.
    - In the 'SIP Entity as Destination' section, click on **Select**.
    - The SIP Entity List page opens.
      Select the entry of the Communication Manager added in **Section 3.3.1** and click on **Select**
    - The selected SIP Entity displays on the Routing Policy Details page.
    - Click on **Commit** to save.

Shown below is the updated screen for the sample configuration which includes the list of dial patterns for any extension numbers that SIP users will dial to reach stations on the Communication Manager Access Element.

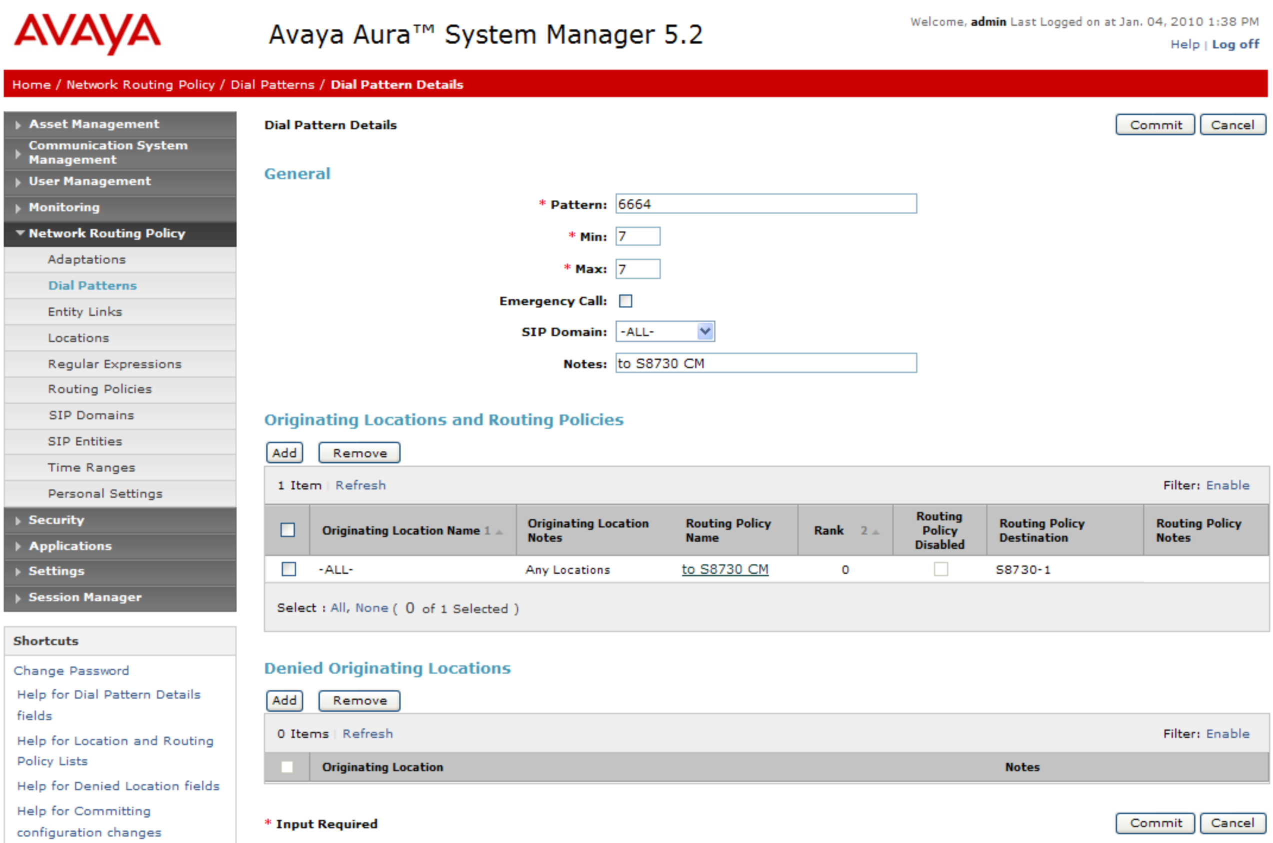


3.3.4. Define Dial Plan for calls to Avaya Aura™ Communication Manager Access Element

- Expand Network Routing Policy
  - o Dial Patterns
    - Click **New**
    - In the 'General' section, under *Pattern* add the numbers that SIP users will dial to reach other extensions on the Communication Manager Access Element. Under *Min* enter the minimum number digits that must to be dialed. Under *Max* enter the maximum number digits that may be dialed.
    - Under SIP Domain drop-down, select the SIP Domain added in **Section 3.1** or select "All" if the system can accept incoming call from all SIP domains.
    - Under *Notes* add a brief description.

- In the 'Originating Locations and Routing Policies' section click on **Add**
  - The 'Locations and Routing Policy List' page opens.
  - Under Locations, select the desired location.
- Under Routing Policies, select the one defined for Communication Manager in **Section 3.3.2** and click on **Select**.

Shown below is the updated screen for one of the dial patterns in the sample configuration.

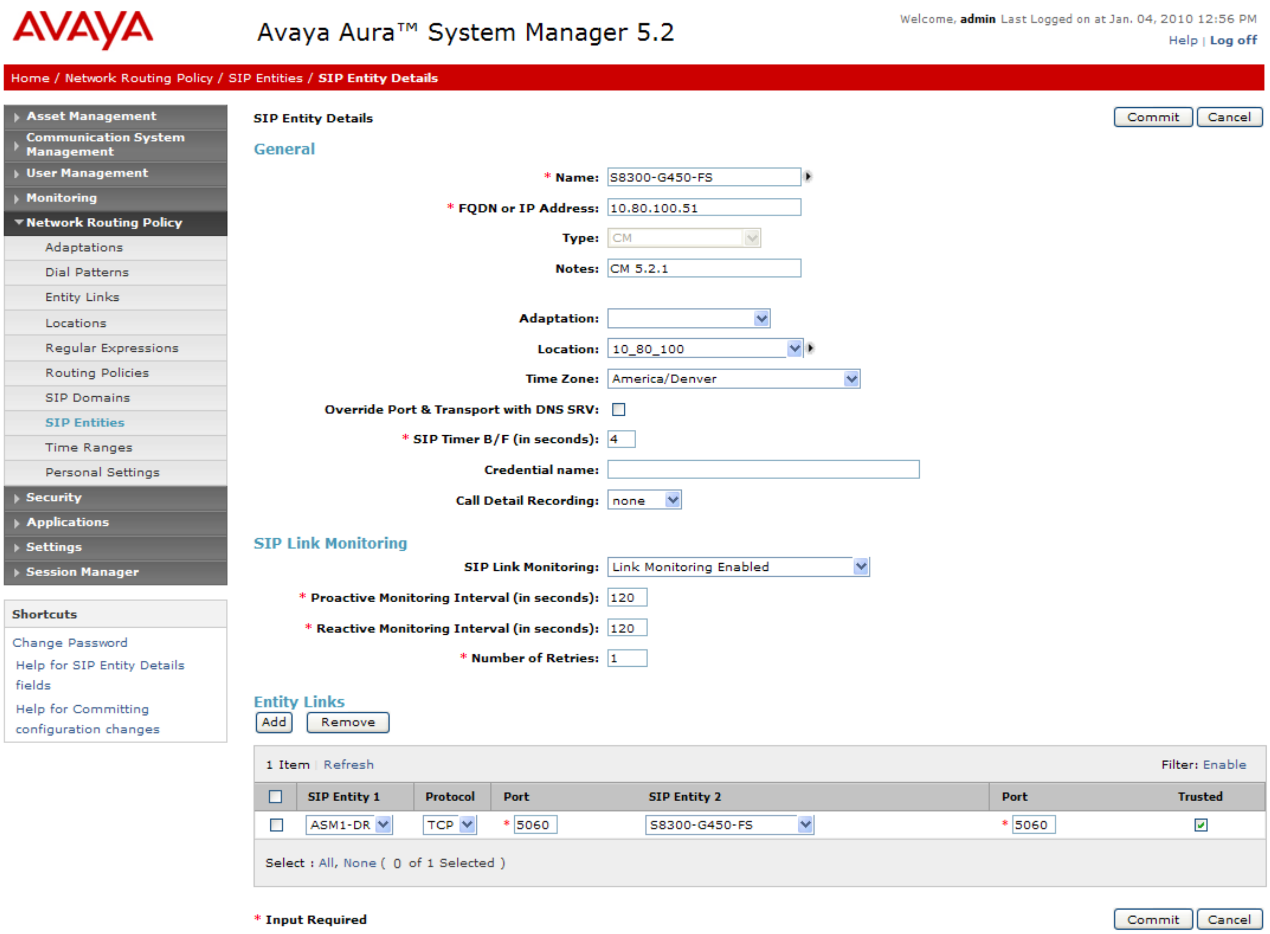


3.4. Add Avaya Aura™ Communication Manager Feature Server

The following section captures relevant screens for configuring Avaya Aura™ Communication Manager Feature Server applicable for the sample configuration.

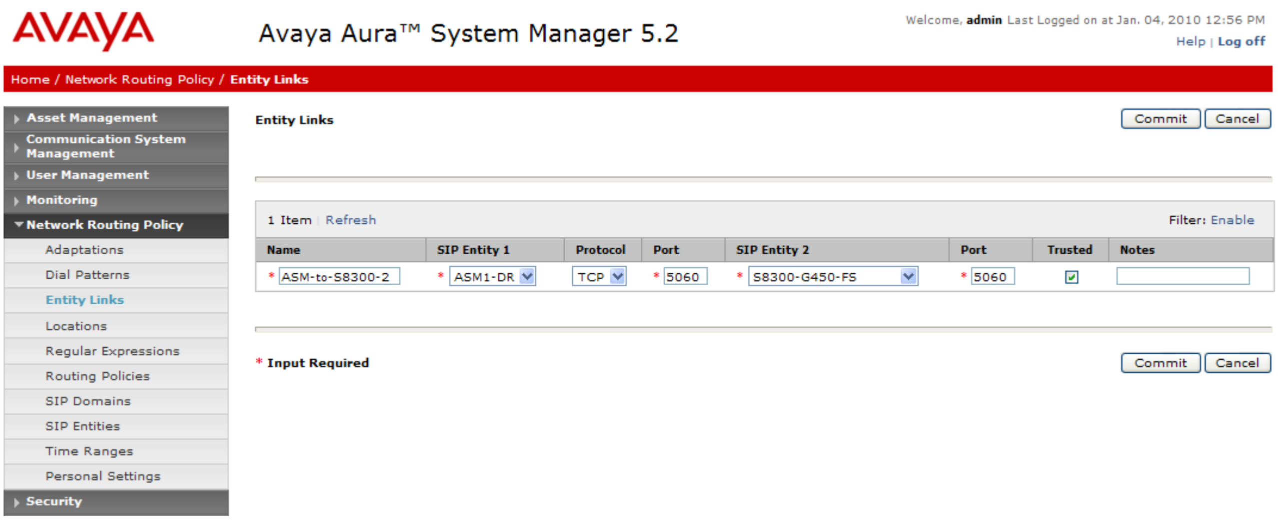
3.4.1. Define a SIP Entity for Avaya Aura^{TM} Communication Manager Feature Server

The following screen shows addition of Communication Manager Feature Server. The IP address used is that of the S8300C server.



3.4.2. Define Entity Link for Avaya Aura^{TM} Communication Manager Feature Server

The following screen shows the entity link defined for the Avaya Aura^{TM} Communication Manager Feature Server.

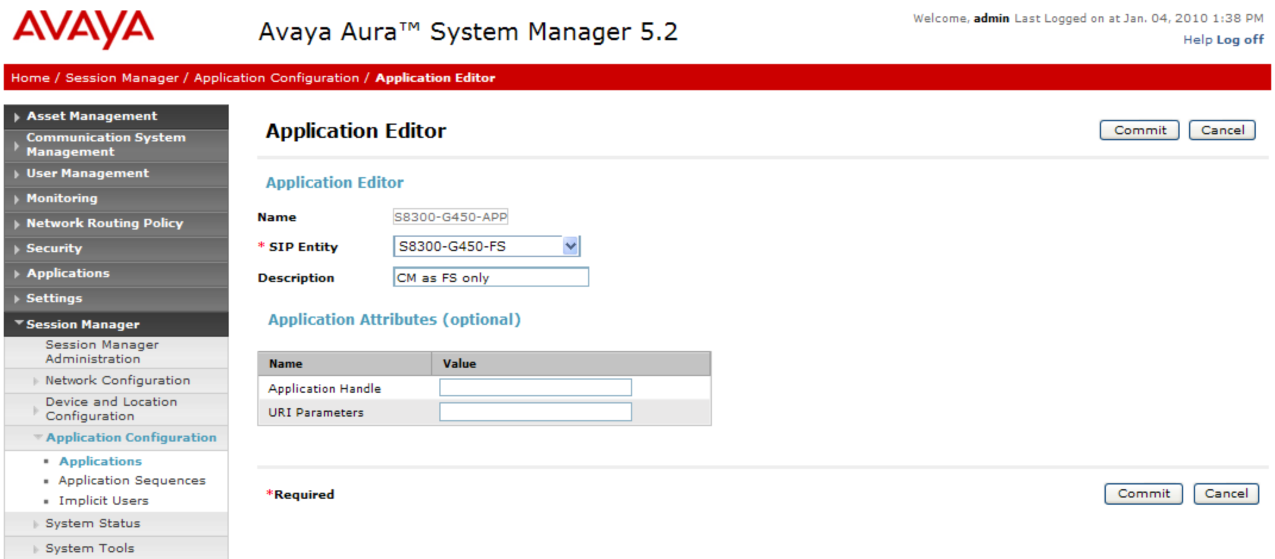


3.4.3. Define Routing Policy for Avaya Aura^{TM} Communication Manager Feature Server

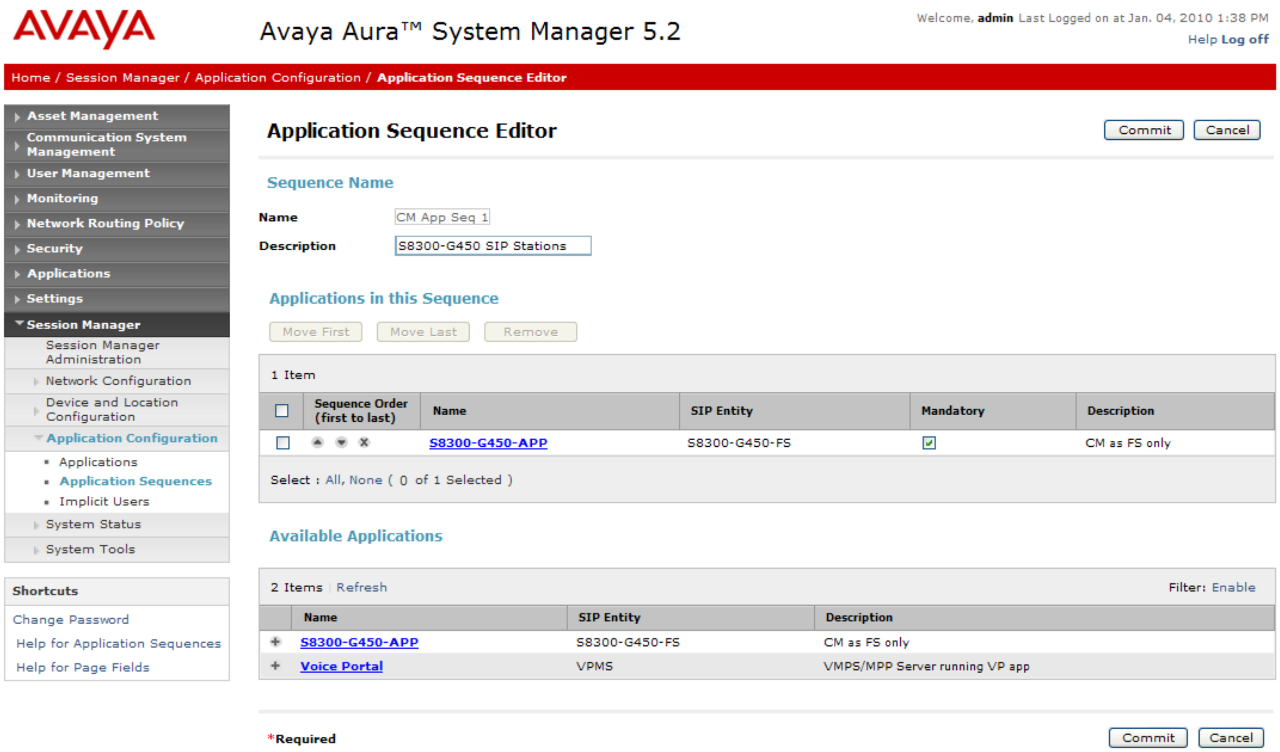
Since the SIP users are registered on Session Manager, a routing policy does not need to be defined for the Communication Manager Feature Server.

3.4.4. Define Application Sequence for Avaya Aura^{TM} Communication Manager Feature Server

Define an application for the Avaya Aura^{TM} Communication Manager Feature Server as shown below:



Second, define an application sequence for the Avaya Aura^{TM} Communication Manager Feature Server as shown below:

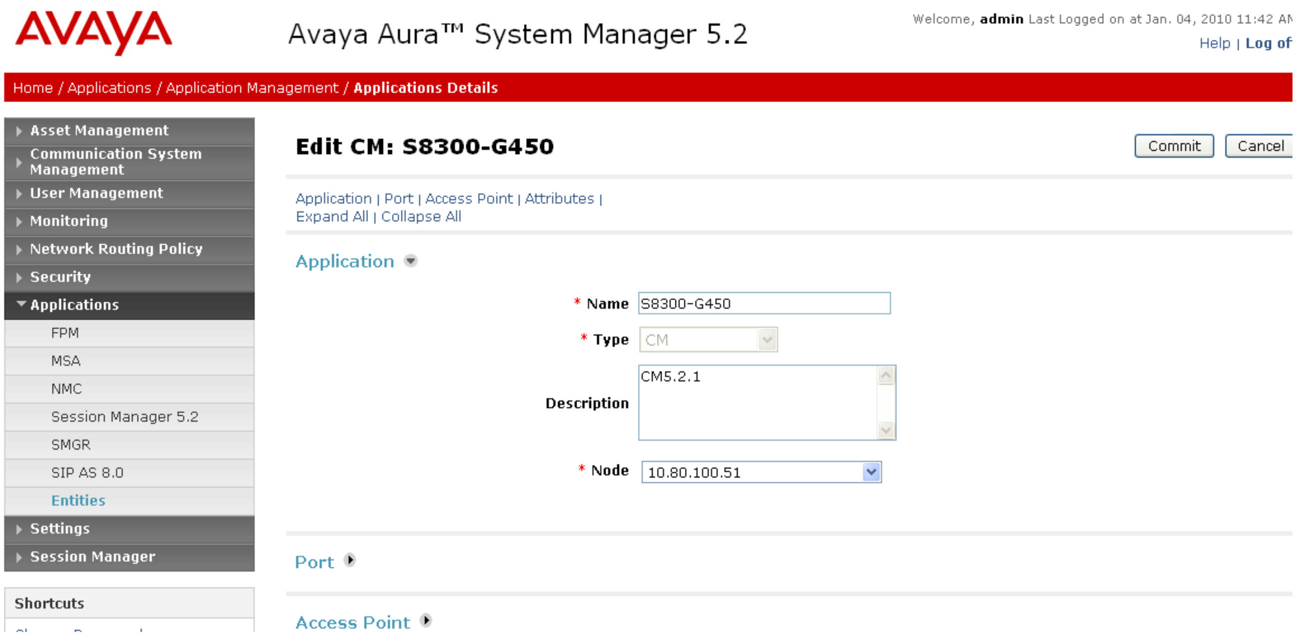


3.4.5. Define Avaya Aura^{TM} Communication Manager Feature as an Administrable Entity

Before adding SIP users, the Avaya Aura^{TM} Communication Manager Feature Server must also be added to System Manager as an administrable entity. This action allows System Manager to access Communication Manager over its administration interface similar to how other administration tools such as Avaya Site Administrator access Communication Manager. Using this administration interface, System Manager will notify the Communication Manager Feature Server when new SIP users are added.

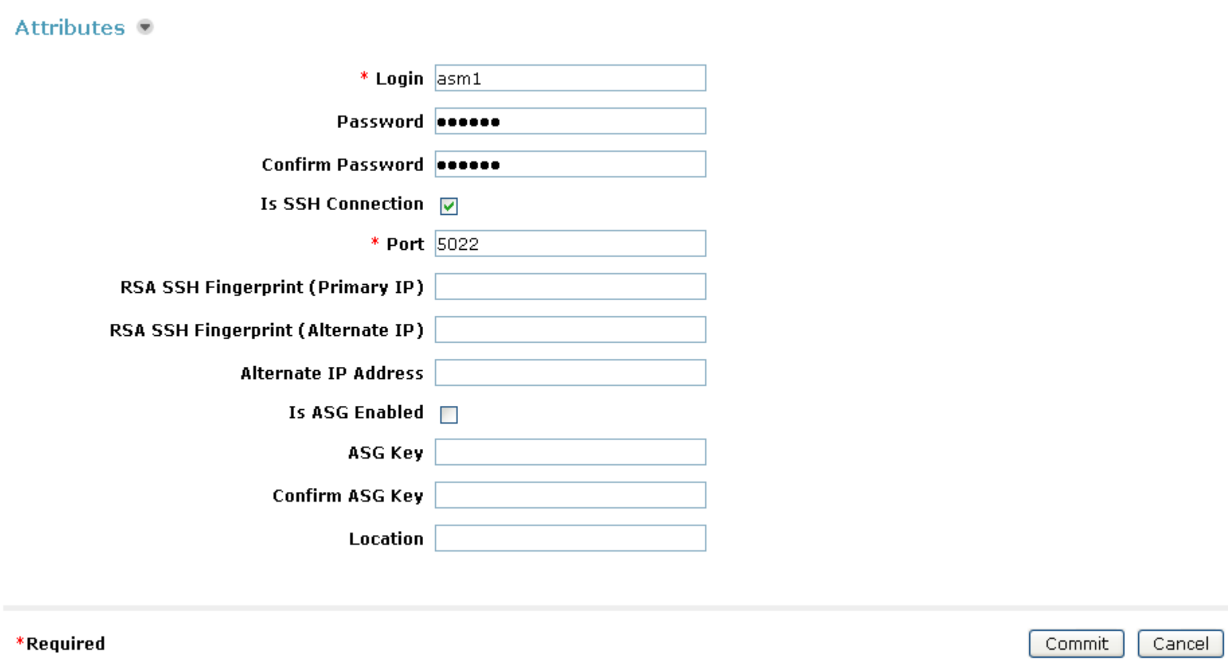
To define the Avaya Aura^{TM} Communication Manager Feature Server as an administrable entity,

- Expand Applications
  - o Entities -> Applications
    - Click **New**
    - Under *Name,* enter an identifier for the Communication Manager Feature Server.
    - Under *Type* drop-down menu, select CM.
    - Under *Node,* enter the IP address of the administration interface for the Feature Server as shown below:



Defining the Avaya Aura^{TM} Communication Manager Feature Server as an administrable entity (continued):

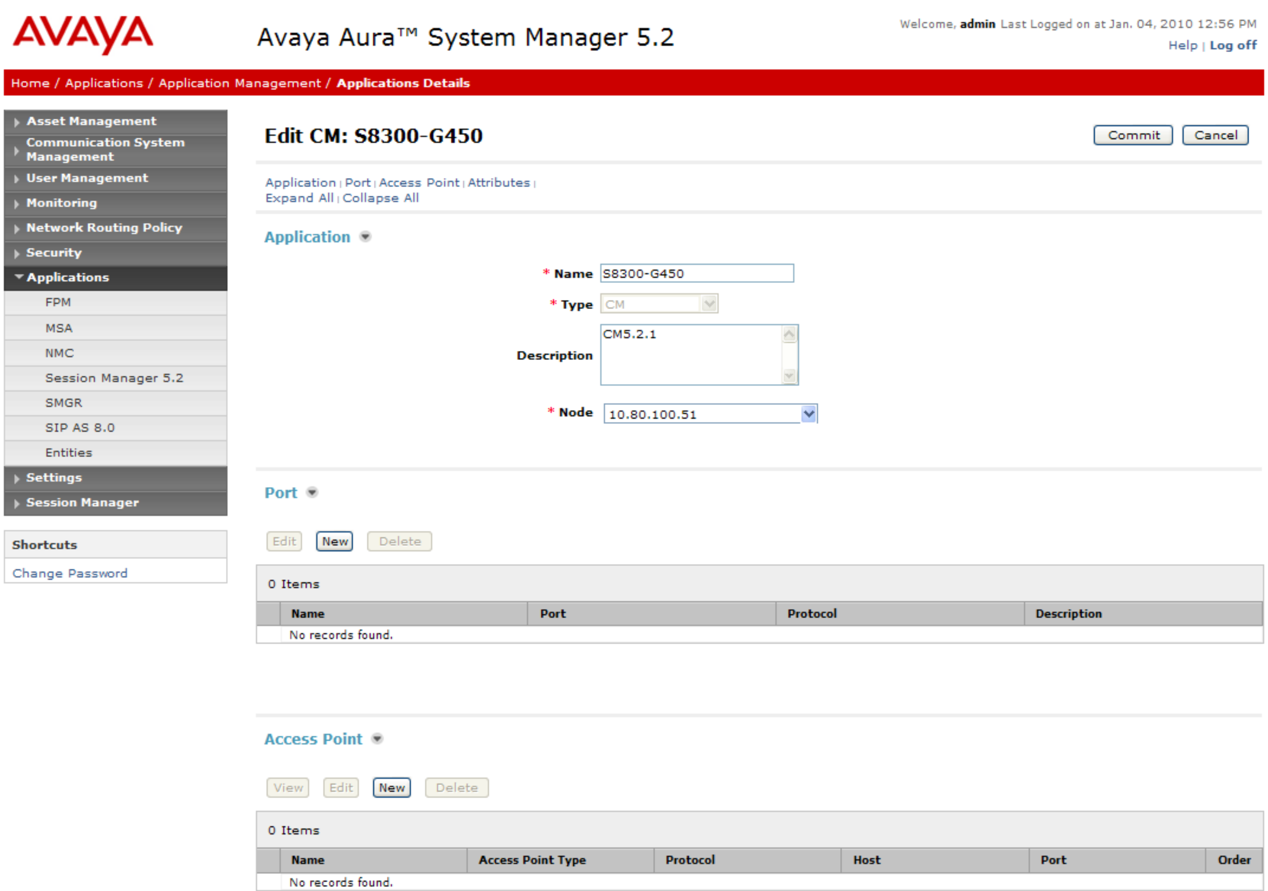
- o Entities - Attributes
  - Under *Login and Password,* enter the login and password used for administration access to the Feature Server.
  - Select SSH access.
  - Under *Port,* enter the port number for the administration interface of 5022 as shown below:



Defining the Avaya Aura^{TM} Communication Manager Feature Server as an administrable entity (continued):

- o Entities – Port
- o Entities – Access Point

Although the port number for the administration interface is defined under the Attribute tab, no additional data is needed for either the Port or Access Point tabs as shown below:



3.4.6. Add SIP Users

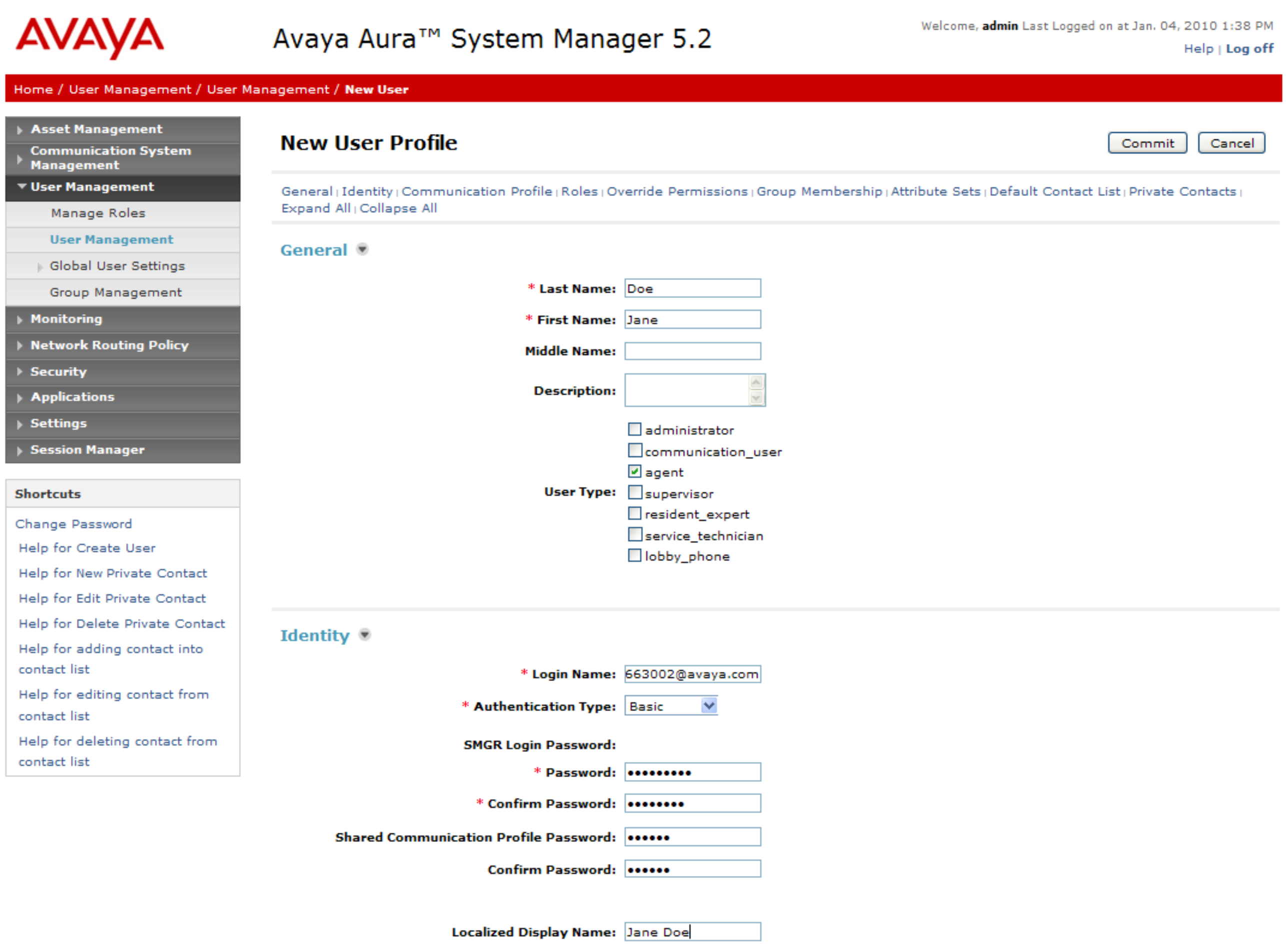
Add SIP users corresponding to the 96XX SIP stations defined in **Section 2.7.** Alternatively, use the option to automatically generate the SIP stations on Communication Manager Feature Server when adding a new SIP user.

- Expand User Management
  - o Select User Management
    - Click **New**

<u>Step 1</u>: Enter values for the following required attributes for a new SIP user in the **General** and **Identity** sections of the new user form.

- **Last Name:** enter last name of user
- **First Name:** enter first name of user
- **Login Name:** enter extension no.@sip domain defined in **Section 3.1**. This field is primary handle of user.
- **Authentication Type:** select **Basic**
- **SMGR Login Password:** enter password which will be used to log into System Manager application
- **Confirm Password:** repeat value entered above
- **Shared Communication Profile Password:** enter a numeric value which will be used to logon to SIP phone. *Note*: this field must match the Security Code field on the station form defined in **Section 2.7.**
- **Confirm Password:** repeat numeric password

The screen below shows the information when adding a new SIP user to the sample configuration.



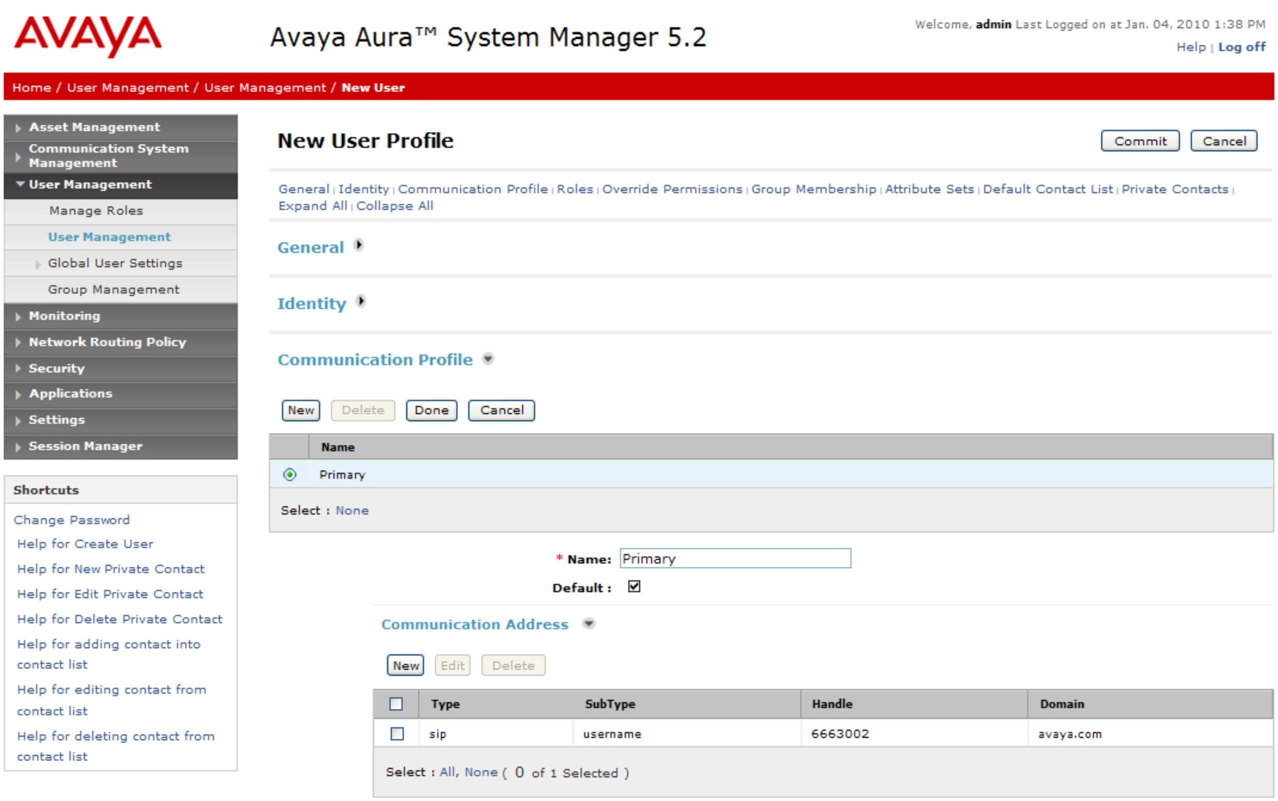
Step 2: Scroll down to the Communication Profile section and select **New** to define a **Communication Profile** for the new SIP user. Enter values for the following required attributes:

- **Name:** enter name of communication profile
- **Default:** enter checkmark to indicate this profile is default profile

Select **New** to define a **Communication Address** for the new SIP user. Enter values for the following required attributes:

- **Type:** select SIP
- **SubType:** select username
- **Handle:** enter extension number
- **Domain:** enter SIP domain defined in **Section 3.1**

The screen below shows the information when adding a new SIP user to the sample configuration.

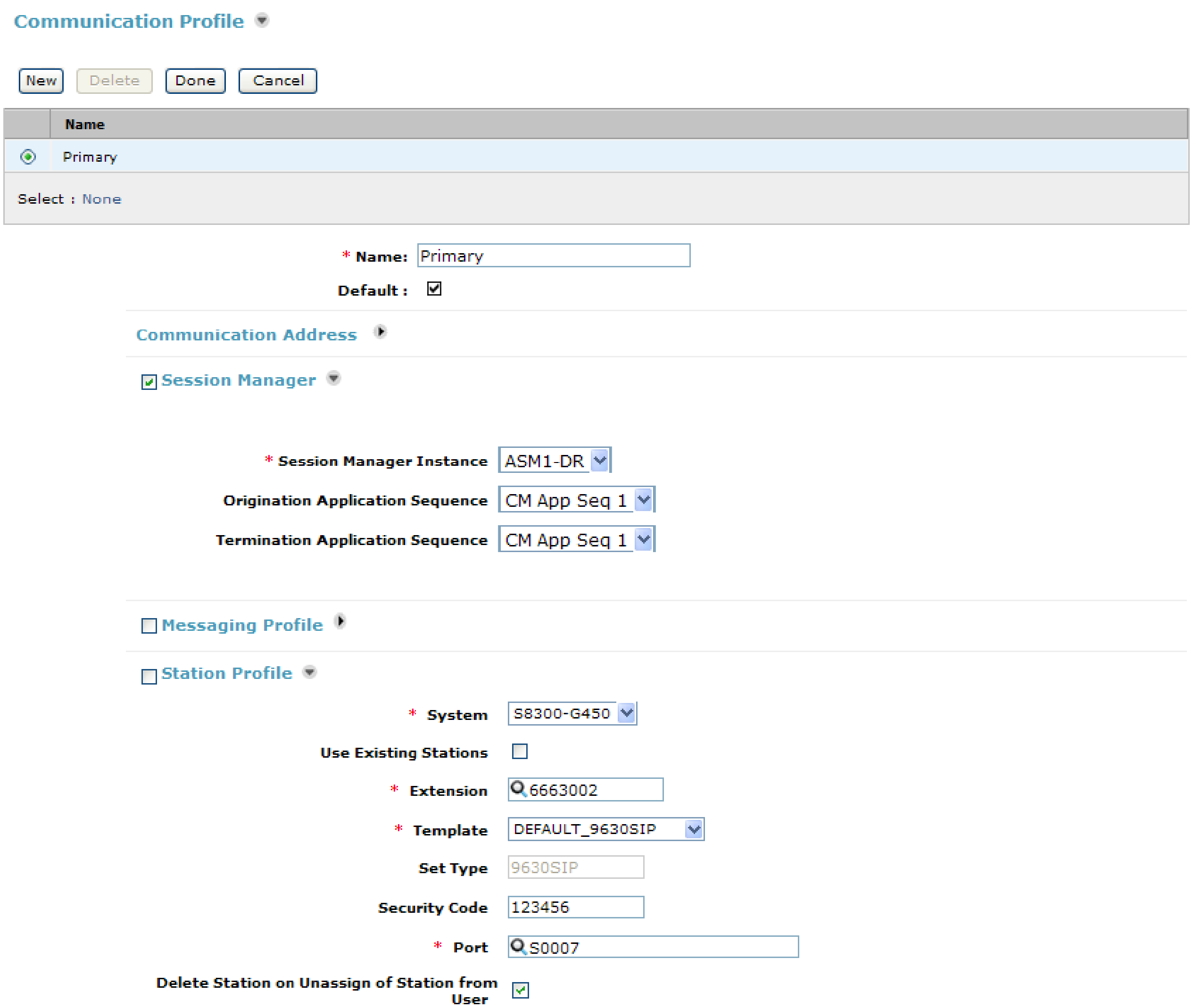


Step 3: Assign the **Application Sequence** defined in **Section 3.4.4** to the new SIP user as part of defining the **SIP Communication Profile**. The **Application Sequence** can be used for both the originating and terminating sequence. Enter values for the following required attributes of the **Station Profile** section:

- **System:** select the SIP Entity of the Communication Manager Feature Server defined in **Section 3.4.1** from menu

- **Use Existing Stations:** enter checkmark if station was already defined. Else, station will automatically be created.
- **Extension**: enter extension number
- **Template:** select template for type of SIP phone
- **Security Code:** enter numeric value which will be used to logon to SIP phone.
  Note: this field must match the value entered for the **Shared Communication Profile Password** field
- **Port**: select port number from the list for the selected template
- **Delete Station on Unassign of Station**: enter checkmark to automatically delete station when **Station Profile** is un-assigned from user.

The screen below shows the information when adding a new SIP user to the sample configuration.



4. Configuring Avaya Aura^{TM} Communication Manager Access Element

This section describes the administration of Communication Manager Access Element using a System Access Terminal (SAT). Some administrative screens are not shown in this section, as they might be similar to **Section 2.**

- Verify System Capabilities and Communication Manager Licensing
- Administer IP network region
- Administer IP node names
- Administer SIP trunk group and signaling group
- Administer route patterns
- Administer numbering plan

After completing these steps, the "**save translations**" command should be performed.

4.1. Verify System Capabilities and Licensing

This section describes the procedures to verify the correct system capabilities and licensing have been configured. If there is insufficient capacity or a required feature is not available, contact an authorized Avaya sales representative to make the appropriate changes.

4.1.1. SIP Trunk Capacity Check

Use the "**display system-parameters customer-options"** command to verify that an adequate number of SIP trunk members are administered for the system. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column. The difference between the two values needs to be greater than or equal to the desired number of simultaneous SIP trunk connections.

4.1.2. AAR/ARS Routing Check

Verify that **ARS** is enabled (on page 3 of system-parameters customer options).

4.1.3. Configure Trunk-to-Trunk Transfers

Use the "**change system-parameters features**" command to enable trunk-to-trunk transfers.

4.2. Configure Codec Type

Issue the **change ip-codec-set n** command where **n** is the next available number. Enter the following values:

- Enter "**G.711MU**" and **"G.729"** as supported types of Audio Codecs
- Silence Suppression: Retain the default value "**n**".

- Frames Per Pkt: Enter “**2**”.
- Packet Size (ms): Enter “**20**”.
- Media Encryption: Enter the value based on the system requirement. For the sample configuration, “none” was used.

```
change ip-codec-set 1                                      Page   1 of   2
                          IP Codec Set

    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n         2        20
 2: G.729             n         2        20
 3:

     Media Encryption
 1: none
```

4.3. Set IP Network Region

Using the **change ip-network-region 1** command, set the **Intra-region IP-IP Direct Audio**, and **Inter-region IP-IP Direct Audio** fields to “**yes**”. For the **Codec Set** enter the corresponding audio codec set configured in **Section 4.1**. Set the **Authoritative Domain** to the correct SIP domain for the configuration.

```
change ip-network-region 1                                 Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: avaya.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 16585
```

4.4. Add Node Names and IP Addresses

Using the **change node-names ip** command, add the node-name and IP for the CLANs and the Session Manager, if not already previously added. Note the node names of the CLANs which will later be used to configure the SIP trunks between the Avaya G650 and the Session Manager.

```
change node-names ip                                       Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
8730-1              10.80.111.11
8730-2              10.80.111.12
ASM1                10.80.100.24
ASM2                10.80.100.26
CLAN-1              10.80.111.16
CLAN-2              10.80.111.17
```

4.5. Configure SIP Signaling Group and Trunk Group

4.5.1. Create a Signaling Group for SIP Trunk to Avaya Aura™ Session Manager

Use the **add signaling-group n** command, where "n" is an available signaling group number to create a SIP trunk to the Session Manager. In the sample configuration, trunk group "10" and signaling group "10" were used to connect to Avaya Aura™ Session Manager. Fill in the indicated fields as shown below. Default values can be used for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tcp[3]"
- **IMS Enabled:** "n"
- **Near-end Node Name:** C-LAN node name from **Section 4.4**.
- **Far-end Node Name:** Session Manager node name from **Section 4.4**.
- **Near-end Listen Port:** "5060"
- **Far-end Listen Port:** "5060"
- **Far-end Domain:** enter domain name defined in IP Network Region for **Authoritative Domain** field. **See Section 4.3**
- **DTMF over IP:** "rtp-payload"
- **Session Establishment Timer:** "3" [4]

```
add signaling-group 10                                      Page   1 of   1
                                SIGNALING GROUP

Group Number: 10             Group Type:  sip
                        Transport Method: tcp
  IMS Enabled? n
     IP Video? n

    Near-end Node Name: CLAN-2                 Far-end Node Name: ASM1
 Near-end Listen Port: 5060                 Far-end Listen Port: 5060
                                       Far-end Network Region:
Far-end Domain: avaya.com
                                           Bypass If IP Threshold Exceeded? n
       DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                   IP Audio Hairpinning? n
         Enable Layer 3 Test? n                   Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n          Alternate Route Timer(sec): 6
```

[3] TCP was used for the sample configuration. However, TLS would typically be used in production environments.

[4] If any call originating from the SIP phone is not expected to be answered within 3 minutes such would happen if the call is made to a VDN and agents are not available within 3 minutes, this value may need to be increased.

4.5.2. Add a SIP Trunk Group to Connect to Avaya Aura™ Session Manager

Add the corresponding trunk group controlled by this signaling group via the **add trunk-group n** command, where "n" is an available trunk group number and fill in the indicated fields.

- **Group Type:** "sip"
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** "tie"
- **Signaling Group:** The number of the signaling group added in **Section 4.5.1**
- **Number of Members:** The number of members in the SIP trunk to be allocated to calls routed to Session Manager (must be within the limits of the total number of trunks configured in **Section 4.1.1**).

Once the add command is completed, trunk members will be automatically generated based on the value in the **Number of Members** field.

```
add trunk-group 10                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 10                  Group Type: sip          CDR Reports: y
  Group Name: SIP trunk to ASM1           COR: 1        TN: 1        TAC: #10
Direction: two-way        Outgoing Display? n
 Dial Access? n                                  Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 10
                                                     Number of Members: 10
```

On page 2, set the **Preferred Minimum Session Refresh Interval** to 1200. Note: to avoid extra SIP messages, all SIP trunks connected to Session Manager should be configured with a minimum value of 1200.

```
add trunk-group 10                                            Page   2 of  21
                                Group Type: sip
TRUNK PARAMETERS

     Unicode Name: auto
                                            Redirect On OPTIM Failure: 5000

            SCCAN? n                                 Digital Loss Group: 18
                      Preferred Minimum Session Refresh Interval(sec): 1200
```

On page 3, set **Numbering Format** to be *public*. Use default values for all other fields.

```
add trunk-group 10                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y

                     Numbering Format: public
                                              UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n

 Show ANSWERED BY on Display? y
```

4.6. Configure Route Pattern

Use the “**add route-pattern X**” command, when **X** is an available number to define a route pattern for routing calls over the SIP trunk group defined **in Section 4.5** to Session Manager**.** In the sample configuration, route pattern 10 was created as shown below:

```
add route-pattern 10                                      Page   1 of  3

                    Pattern Number: 10  Pattern Name: SIP to ASM1
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                          DCS/ IXC
    No          Mrk Lmt List Del  Digits                            QSIG
                             Dgts                                   Intw
 1: 10   0                                                           n   user
 2:                                                                  n   user
 3:                                                                  n   user
```

4.7. Administer Numbering Plan

4.7.1. Administer Uniform Dialplan

Use the “**change uniform-dialplan x**” command, where **x** is the first digit of the extension numbers used for SIP stations in the system.

In the sample configuration, extensions starting with “666-3XXX” are used for extensions associated with the 96XX SIP phones.

```
change uniform-dialplan 6                                    Page   1 of  2
                            UNIFORM DIAL PLAN TABLE
                                                            Percent Full: 0
  Matching                        Insert              Node
  Pattern       Len Del           Digits      Net Conv Num
 6663            7   0                        aar  n
 6665000         7   0                        aar  n
 777             7   0                        aar  n
 778             7   0                        aar  n
                                                   n
```

4.7.2. Administer AAR analysis

This section provides the configuration of the AAR pattern used in the sample configuration for routing calls between Communication Manager Access Element and SIP users registered to Session Manager.

Note that other methods of routing may be used.

Use the “**change aar analysis x**” command where **x** is the first digit of the extension numbers used for SIP stations in the system.

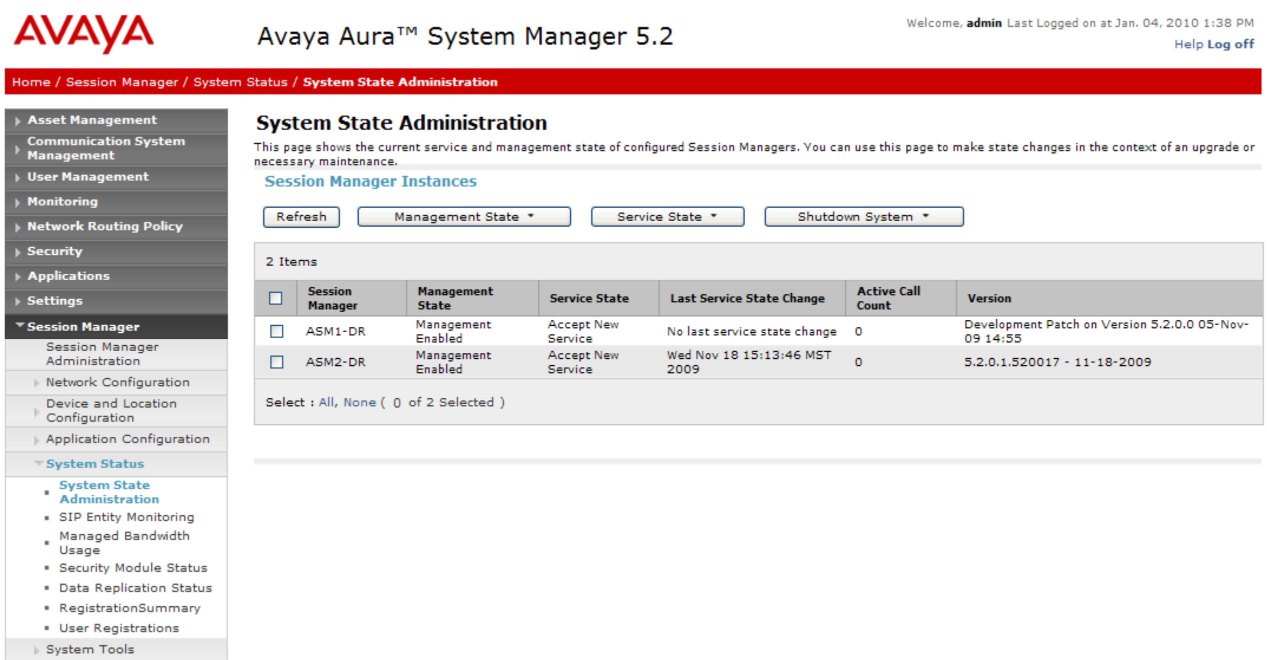
```
change aar analysis 6                                       Page   1 of  2
                          AAR DIGIT ANALYSIS TABLE
                                 Location:  all        Percent Full:    1

          Dialed            Total     Route    Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    6                     7    7     10        aar          n
    6663                  7    7     10        aar          n
    6665000               7    7     20        aar          n
    777                   7    7     20        lev0         n
    778                   7    7     30        aar          n
    8                     7    7     999       aar          n
    9                     7    7     999       aar          n
```

5. Verification Steps

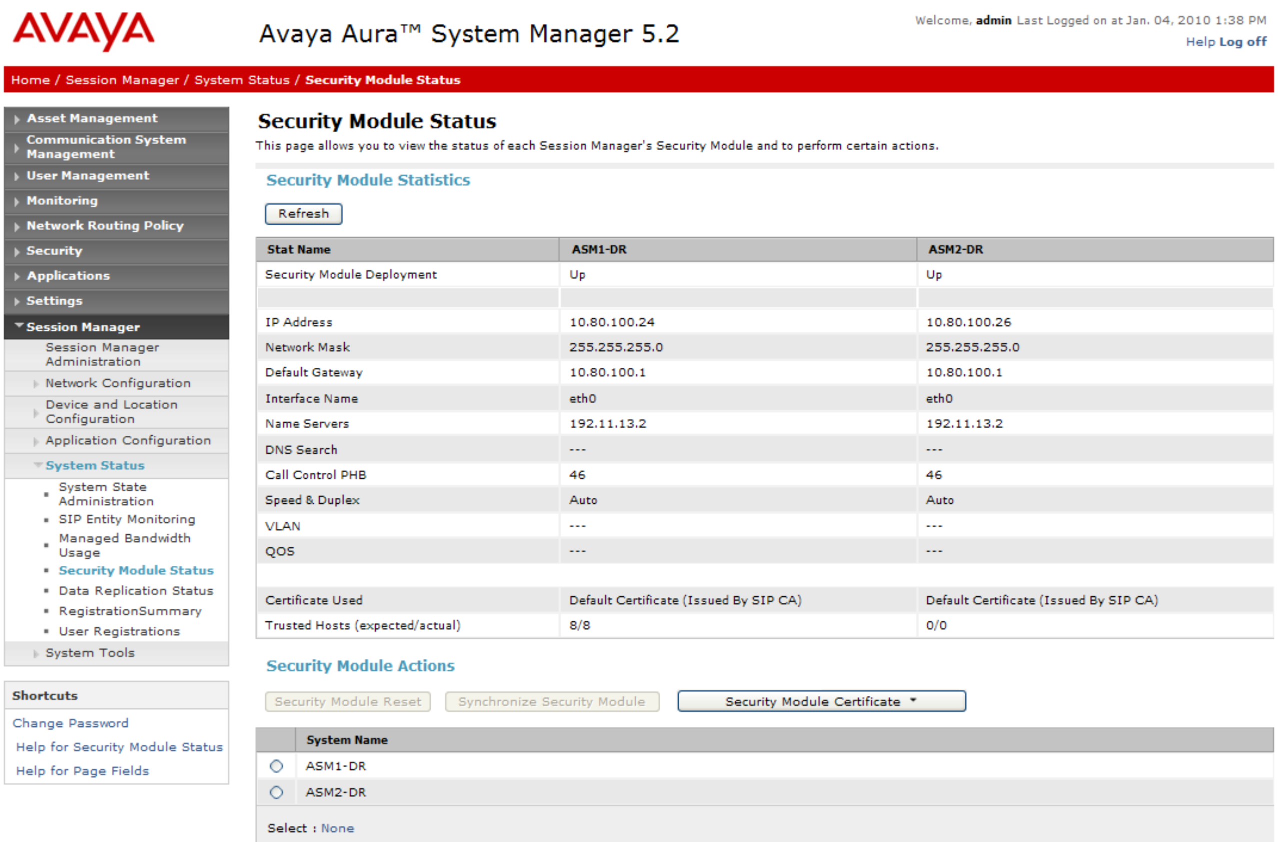
5.1. Verify Avaya Aura™ Session Manager Configuration

5.1.1. Verify Avaya Aura™ Session Manager is Operational

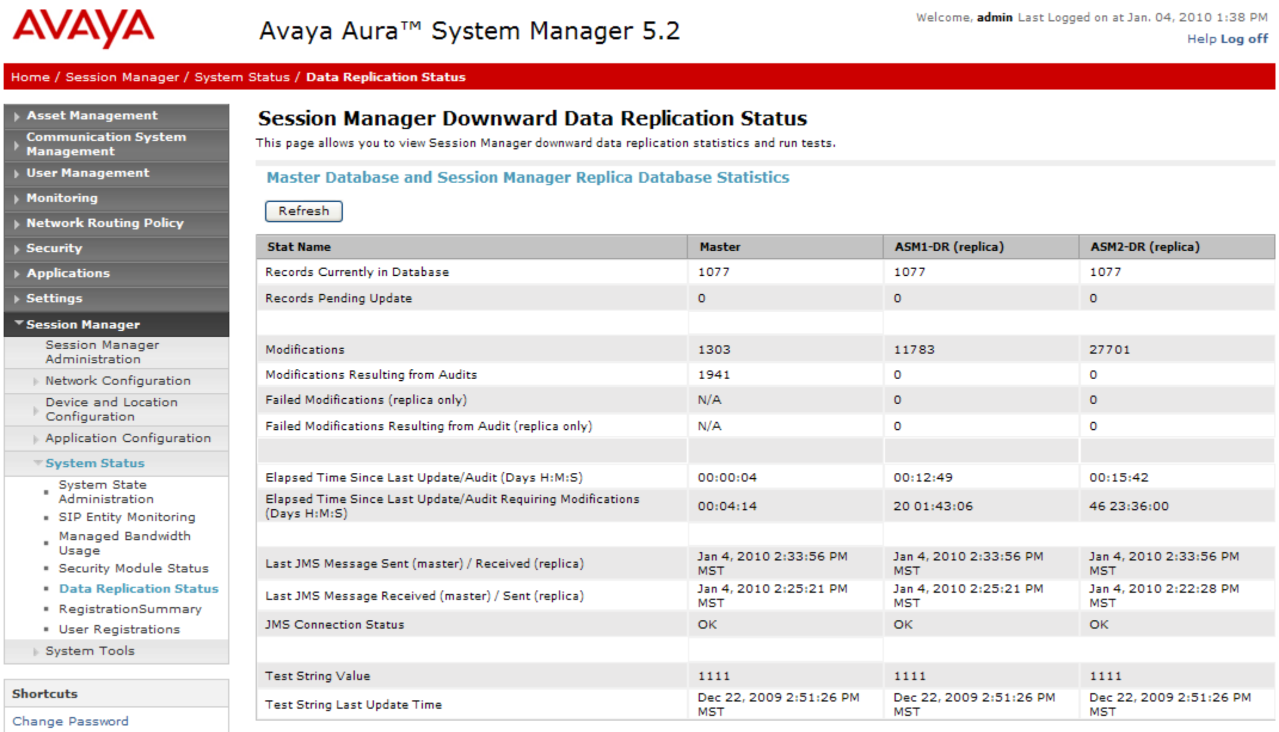
Verify the overall system status for the specific Session Manager as shown below:



Verify the status of the Security Module (SM 100 card) for the specific Session Manager as shown below:

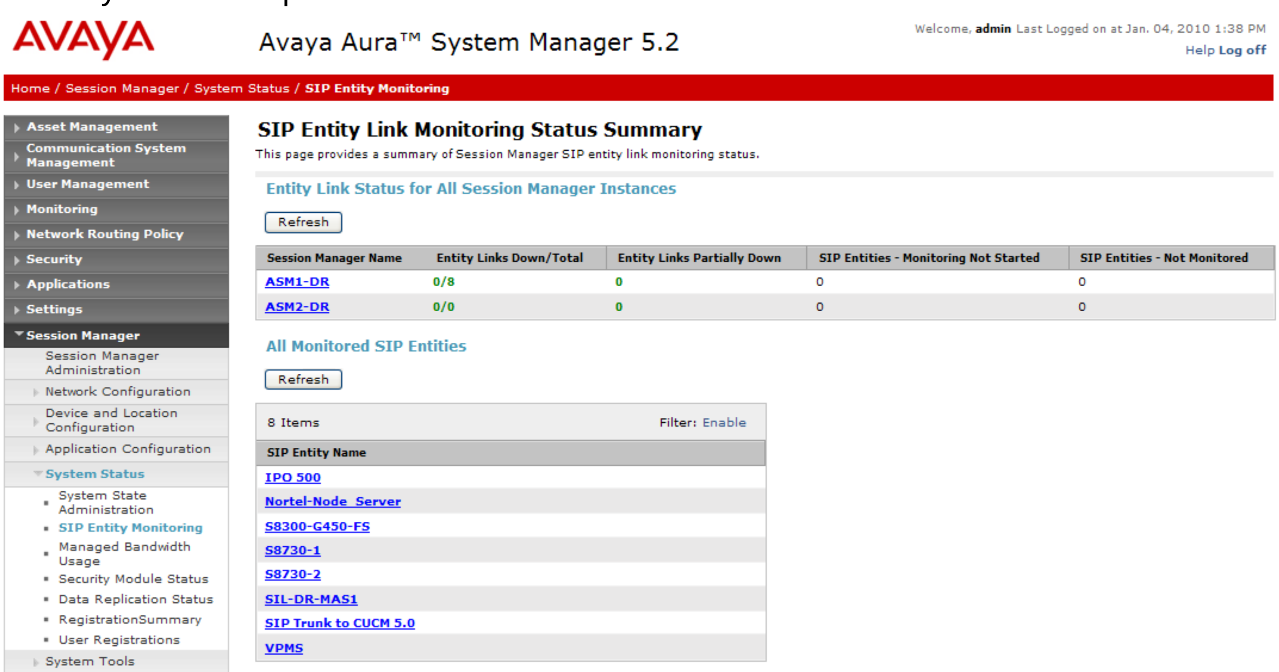


Finally, verify the data replication status is operational as shown below:

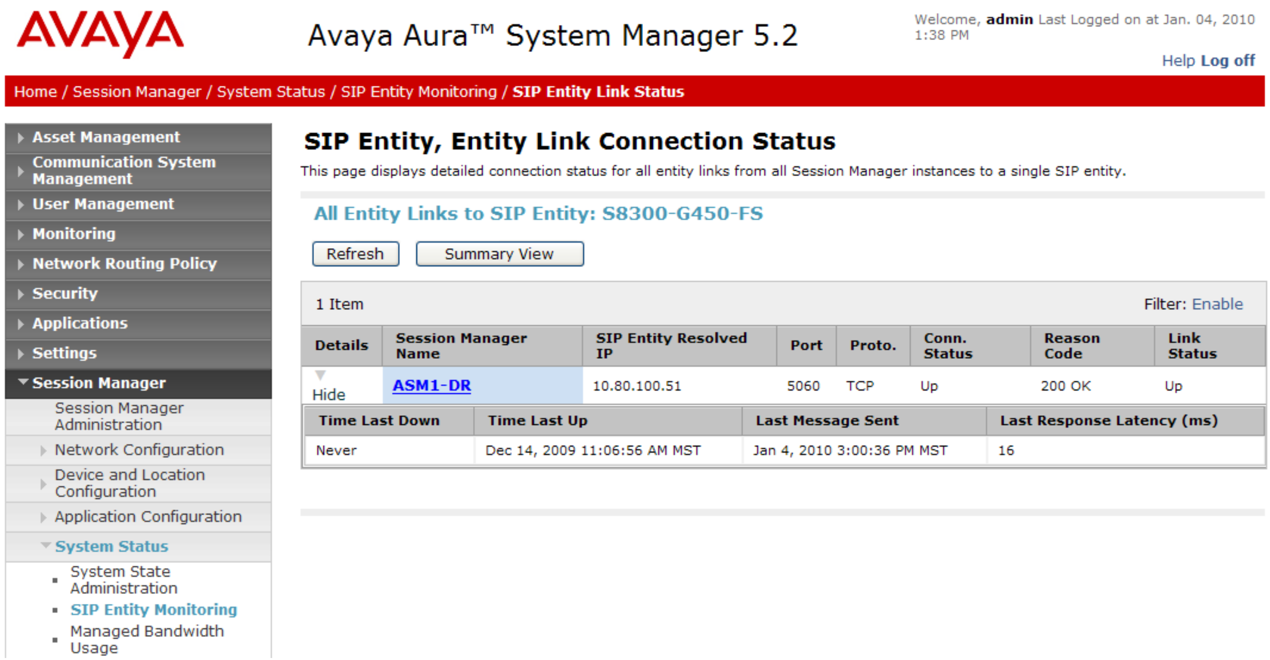


5.1.2. Verify SIP Link Status

Expand the Session Manager menu on the left and click SIP Entity Monitoring. Verify all SIP Entity Links are operational as shown below:

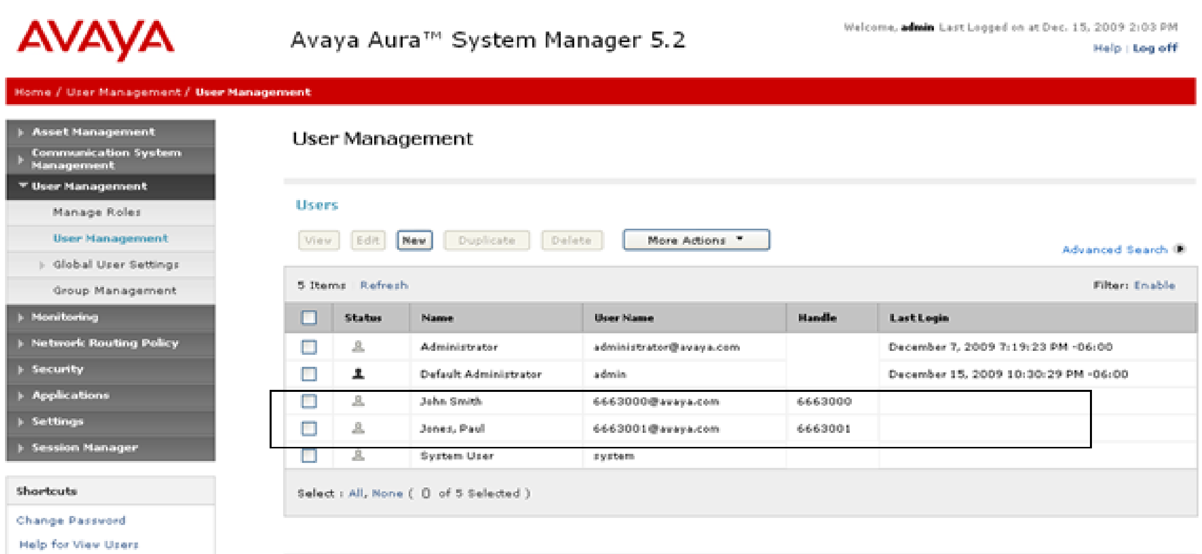


Select the corresponding SIP Entity for the Communication Manager Feature Server and verify the link is up as shown below:

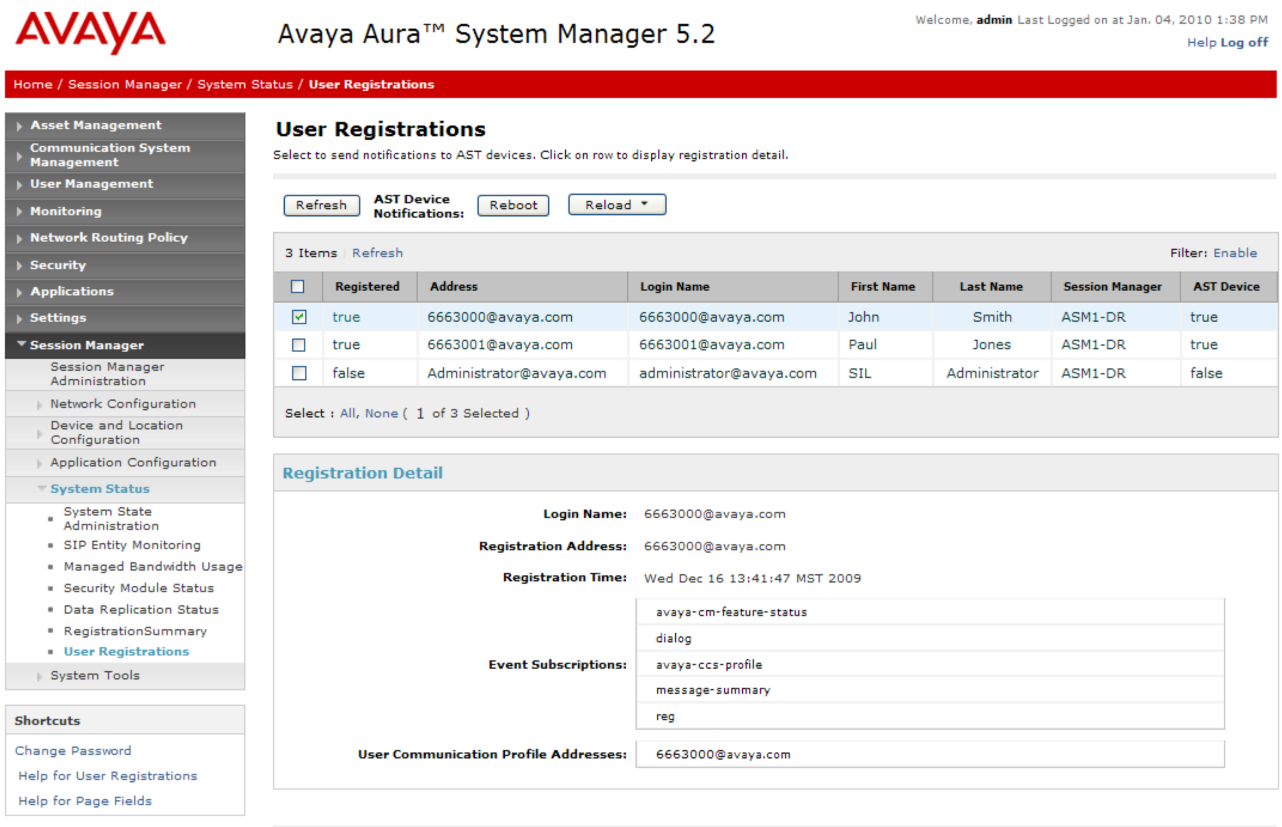


5.1.3. Verify Registrations of SIP Endpoints

Verify SIP users have been created in the Session Manager. In the sample configuration, two SIP users were created as shown in the highlighted area below:



Verify the SIP endpoints have successfully registered with the Session Manager as shown below:



5.2. Verify Avaya Aura™ Communication Manager Feature Server Configuration

Verify the status of the SIP trunk group by using the "**status trunk n**" command, where "**n**" is the trunk group number administered in **Section 2.5.** Verify that all trunks are in the "in-service/idle" state as shown below:

```
status trunk 10
                             TRUNK GROUP STATUS
Member    Port      Service State      Mtce Connected Ports
                                       Busy
0010/001 T00006   in-service/idle     no
0010/002 T00007   in-service/idle     no
0010/003 T00008   in-service/idle     no
0010/004 T00009   in-service/idle     no
0010/005 T00014   in-service/idle     no
0010/006 T00015   in-service/idle     no
0010/007 T00043   in-service/idle     no
0010/008 T00044   in-service/idle     no
0010/009 T00045   in-service/idle     no
0010/010 T00046   in-service/idle     no
```

Verify the status of the SIP signaling groups by using the "**status signaling-group n**" command, where "**n**" is the signaling group number administered in **Section 2.4.** Verify the signaling group is "in-service" as indicated in the **Group State** field shown below:

```
status signaling-group 10
                         STATUS SIGNALING GROUP

        Group ID: 10                                Active NCA-TSC Count: 0
      Group Type: sip                                Active CA-TSC Count: 0
  Signaling Type: facility associated signaling
     Group State: in-service
```

Use the Communication Manager SAT command, '**list trace tac #'**, where **tac #** is the trunk access code defined in **Section 2.5** to trace trunk group activity for the SIP trunk between the Session Manager and the Communication Manager Feature Server as shown below:

```
list trace tac #10                                                        Page   1
                                LIST TRACE
time            data

11:01:01     Calling party station    6663000 cid 0x9e
11:01:01     Calling Number & Name 6663000 John Smith
11:01:01     active station    6663000 cid 0x9e
11:01:07     dial 6664000 route:UDP|AAR
11:01:07     term trunk-group 10    cid 0x9e
11:01:07     dial 6664000 route:UDP|AAR
11:01:07     route-pattern  10 preference 1  cid 0x9e
11:01:07     seize trunk-group 10 member 7  cid 0x9e
11:01:07     Calling Number & Name NO-CPNumber NO-CPName
11:01:07     Setup digits 6664000
11:01:07     Calling Number & Name 6663000 John Smith
11:01:07     Proceed trunk-group 10 member 7  cid 0x9e
11:01:07     Alert trunk-group 10 member 7  cid 0x9e
11:01:07     G711MU ss:off ps:20
```

Use the Communication Manager SAT command, '**list trace station xxx'**, where **xxx** is the extension number of the 96XX SIP telephone as shown below:

```
list trace station 6663000                                                  Page   1
                                LIST TRACE
time            data

11:03:30     active station    6663000 cid 0x9f
11:03:33     dial 6664000 route:UDP|AAR
11:03:33     term trunk-group 10    cid 0x9f
11:03:33     dial 6664000 route:UDP|AAR
11:03:33     route-pattern  10 preference 1  cid 0x9f
11:03:33     seize trunk-group 10 member 8  cid 0x9f
11:03:33     Calling Number & Name NO-CPNumber NO-CPName
11:03:33     Setup digits 6664000
11:03:33     Calling Number & Name 6663000 John Smith
11:03:33     Proceed trunk-group 10 member 8  cid 0x9f
11:03:33     Alert trunk-group 10 member 8  cid 0x9f
11:03:33     G711MU ss:off ps:20
             rgn:1 [10.80.111.13]:9808
             rgn:1 [10.80.100.53]:2052
11:03:33     xoip options: fax:Relay modem:off tty:US  uid:0x5002c
```

5.3. Call Scenarios Verified

Verification scenarios for the configuration described in these Application Notes included the following call scenarios:

- Place a call from a SIP phone registered to Session Manager to an extension on Communication Manger Access Element. Answer the call and verify talkpath.
- Place a call from an extension on the Communication Manger Access Element to a SIP phone registered to Session Manager. Answer the call and verify talkpath.
- Verify that calls can be transferred from a SIP phone registered to Session Manager to an extension on Communication Manager.
- Verify that calls can be transferred from an extension on Communication Manager Access Element to a SIP phone registered to Session Manager.
- Verify that a SIP phone registered to Session Manager can conference in extensions on Communication Manager Access Element.
- Verify extensions on Communication Manager Access Element can conference in SIP phones registered to Session Manager.

6. Acronyms

AAR	Automatic Alternative Routing (Routing on Communication Manager)
ARS	Alternative Routing Service (Routing on Communication Manager)
CLAN	Control LAN (Control Card in Communication Manager)
DCP	Digital Communications Protocol
DNIS	Dialed Number identification Service
DTMF	Dual Tone Multi Frequency
FQDN	Fully Qualified Domain Name (hostname for Domain Naming Resolution)
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPSI	IP-services interface (Control Card in Communication Manager)
LAN	Local Area Network
MRCP	Media Resource Control Protocol
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SAT	System Access Terminal (Communication Administration Interface)
SIL	Solution Interoperability Lab
SIP	Session Initiation Protocol
SM	Avaya Aura™ Session Manager
SMGR	System Manager (used to configure Session Manager)
SNMP	Simple Network Management Protocol
SRE	SIP Routing Element
SSH	Secure Shell
SSL	Secure Socket Layer
TAC	Trunk Access Code (Communication Manager Trunk Access)
TCP	Transmission Control Protocol
TCP/IP	Transmission Control Protocol/Internet Protocol
TLS	Transport Layer Security
URE	User Relation Element
URL	Uniform Resource Locator
WAN	Wide Area Network
XML	eXtensible Markup Language

7. Conclusion

These Application Notes describe how to configure the Avaya Aura™ Session Manager, Avaya Aura™ Communication Manager Access Element and Avaya Aura™ Communication Manager operating as a Feature Server to support 9600-Series SIP Telephones. Interoperability testing included successfully making bi-directional calls between several different types of endpoints and use of various features including transfer and conference.

8. Additional References

This section references the product documentation relevant to these Application Notes.

Session Manager

1) Avaya Aura™ Session Manager Overview, Doc ID 03-603323, available at http://support.avaya.com.
2) Installing and Administering Avaya Aura™ Session Manager, Doc ID 03-603324, available at http://support.avaya.com.
3) Avaya Aura™ Session Manager Case Studies, dated January 2, 2010, available at http://support.avaya.com
4) Maintaining and Troubleshooting Avaya Aura™ Session Manager, Doc ID 03-603325, available at http://support.avaya.com.

Communication Manager

5) Hardware Description and Reference for Avaya Aura™ Communication Manager (COMCODE 555-245-207) http://support.avaya.com/elmodocs2/comm_mgr/r4_0/avayadoc/03_300151_6/245207_6/245207_6.pdf
6) SIP Support in Avaya Aura™ Communication Manager Running on Avaya S8xxx Servers, Doc ID 555-245-206, May 2009, available at http://support.avaya.com.
7) Administering Avaya Aura™ Communication Manager, Doc ID 03-300509, May 2009, available at http://support.avaya.com.
8) Administering Avaya Aura™ Communication Manager as a Feature Server, Doc ID 03-603479, November 2009, available at http://support.avaya.com

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya Solution & Interoperability Test Lab at interoplabnotes@list.avaya.com